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UNDERWATER POSITION-FIXING USING DIGITAL ACOUSTIC COMMUNICATION TECHNIQUES

by

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A Doctoral Thesis

Submitted in partial fulfilment of the requirements

for the award of

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To my Brother Garry, to whom I dedicate this thesis

ABSTRACT

This thesis describes an intelligent underwater acoustic system that allows the positions of several divers or Remotely Operated Vehicles (ROV) to be tracked in three-dimensional space and to telemeter the co-ordinates to a remote receiver at the surface. The positions are fixed using three randomly deployed seabed transponders that may be described as *intelligent*. The transponders fix their own relative positions and the position of the surface receiver, usually a vessel, by an exchange of coded acoustic pulses. These positions can be related to a differential GPS system at the surface if absolute co-ordinates are required. An underwater acoustic positioning and communication system can provide a vital navigation aid for a diver and surface supervisor. Often underwater positioning systems only provide the surface supervisor with diver's positions, with the diver navigating from voiced instruction via an acoustic or wire link communication. In the system described the divers each know their own position from a wrist-worn computer with a backlit graphical/numerical display. As well as the current position, the display can show the track from the beginning of the dive, the location of the surface vessel and the instantaneous position of the other divers.

The design of an intelligent array of transponders and a diver or vehicle self-navigation system is also described. The use of efficient, powerful embedded microprocessors in the transponders enables signal processing and position fixing algorithms to be programmed. The transponders are capable of operating as either mobile or stationary units and in either master or slave mode.

The positioning system transfers small amounts of data between an array of seabed transponders using an advanced Continuous Phase Shift Keying (CPSK) technique. The data packets transferred between the transponder units are precisely *time stamped*, i.e. the exact time that a packet is transmitted or received, to 100ns timing precision. By measuring the acoustic propagation time of the data packet the transponders can accurately position a mobile transponder relative to the seabed array to centimetric precision.

The intelligent transponders form a subsea acoustic network and the access protocols have been developed to enable the transponders to operate without intervention. Several novel rules of operation ensure that a single Master transponder is assigned to

control the communications protocol, as all transponder units are capable of being the Master.

The positioning algorithms operate on the direct slant range measurements and the position is either calculated by the Master unit and telemetered to the Mobile unit or, when operating in a fast positioning mode, the Mobile unit calculates its own position.

This thesis describes a complete practical design and the development of an underwater positioning system. The hardware design of the transponder modules, communication techniques, network protocols and positioning algorithms are shown in detail. Several major contributions are presented. First, the concept of an intelligent seabed array of transponders, which can resolve priority 'contentions' and allocate array-specific individual identification addresses autonomously. The hardware and software are identical for the entire transponder unit, simplifying the reproducibility; however, the transponders have no 'individuality', and hence the communication protocols ensure that individual transponders can be identified. Second, the embedded hardware implementation and software design of the system presents a truly intelligent solution. Third, the phase modulated communication technique suppresses out-of-band interference, facilitating fast symbol rates. Fourth, the dynamic packet data length minimises the problem of multipath interference. Fifth, the extremely high timing precision for measuring when packets are transmitted and captured, enables range calculations with a repeatable measurement error of less than ±1mm. Finally, the design of novel low-power acoustic marker beacons used for underwater marking of objects, diver shot lines, positioning array baselines and other underwater non-positioning applications is described.

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GLOSSARY

ADC	Analogue to Digital Converter
AGC	Automatic Gain Control
ALAP	Acoustic Link Access Protocol
AUV	Autonomous Underwater Vehicle
BER	Bit Error Rate
CDMA	Code-Division Multiple-Access
CRC	Cyclic Redundancy Check
DAC	Digital to Analogue Converter
EM	Electro-Magnetic
FSF	Frequency-Specific Fading
GLONASS	GLObal NAvigation Satellite System
GPS	Global Positioning Sytem
HDLC	High-level Data Link Control
INS	Inertial Navigation System
KB	K Bits
Kbyte	K Byte
LAAN	Local Area Acoustic Network
LOS	Line Of Sight
IR	Infra Red
IrLAP	Infrared Link Access Protocol
IrDA	Infrared Data Association
ISI	Inter-Symbol Interference
LSB	Least Significant Bit
MSB	Most Significant Bit

N	Noise
NRM	Normal Response Mode
NDM	Normal Disconnect Mode
OP-AMP	Operational Amplifier
PIO	Peripheral Input Output
PP	Point to Point
PM	Point to Multi-point
PTFE	PolyTetraFluoroEthene
ROV	Remotely Operated Vehicle
Rx	Receiver
SDLC	Synchronous Data Link Control
SNR	Signal-to-Noise Ratio
TDMA	Time-Division Multiple-Access
Tx	Transmitter
UAPS	Underwater Acoustic Positioning System
UDI	Underwater Diver Interface
XID	eXchange station IDentification

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CHAPTER ONE

INTRODUCTION

1. INTRODUCTION

1.1 General

The advent of accessible handheld Global Positioning Systems (GPS) receivers enables anyone to know accurately their position to within a few metres anywhere on the surface of the earth, although not underwater. These have become an essential piece of equipment for navigation when mountaineering, sailing, and many other outdoor activities inasmuch as a compass was only a decade ago. GPS receivers are now being integrated into common everyday items, such as cars, mobile phones and tracker systems, and probably the best feature of GPS is that the service is free! The American system is called *NAVSTAR* and a very similar Russian system is known as *GLONASS.*

The GPS is one of the most accurate navigation systems, consisting of a network of 24 low-orbit satellites. Each satellite carries extremely accurate atomic clocks for transmitting timing signals worldwide. The timing accuracy is obtained by using either a rubidium or caesium temperature-compensated atomic clock, which typically drifts by less than 10^{-12} seconds/hour. In addition, the GPS technology enables many other timing and frequency applications[1.1].

The position of a GPS receiver is calculated when signals from a minimum of four satellites are received; however the position accuracy and repeatability can be increased if there is redundancy, i.e. more signals received, enabling a least squares or similar algorithm to be used. The GPS system over the years must have saved thousands of lives by helping people navigate their route in hazardous conditions. The satellites transmit circular-polarised signals at 1.575 GHz (L1), hence in conductive seawater, the Electro-Magnetic (EM) waves do not propagate more than a few metres.

The military has been interested in position fixing underwater for some years, not just for positioning their own vessels, but also for positioning enemy vessels relative to themselves. Locating the position of noise sources underwater has been extremely important to the navies, especially since the introduction of U-Boats (Submarines) at the turn of the 20th century. There are basically two ways of locating the position of an underwater target; passively or actively. Passive localisation requires the target to generate self-noise, such as engine noise in the case of a submarine. Active localisation can position a target that generates no self-noise. An active system transmits a sound and listens to the echoes reflected back to the receiver. The amount of sound reflected back towards the source depends on the Target Strength (TS) and the range to the target.

The two cases above are generally used to localise the position of a target that does not actively want to be positioned. This thesis is not concerned with the above military application of target detection and localisation, such as in Mine Counter Measures (MCM) and Anti-Submarine Warfare (ASW). This thesis is directed towards offering an underwater GPS, for underwater exploration by commercial and sports divers.

When conducting underwater tasks, it is desirable to have some means of co-ordinating the activity of the divers. To fulfil this desire, many systems have been designed, which enable communications from the surface to the diver. These systems are classified as wire-link and wireless, and have been successfully used for underwater communications for many years, although mainly in the military environment. The wire-link system is preferred if an umbilical line is available. The wireless, acoustic systems are chosen when the diver requires a greater degree of freedom, as in the case of SCUBA divers. The ability to communicate to divers from the surface and vice versa enables the surface vessel to co-ordinate the underwater activities. However, the co-ordination is performed blind, as the surface vessel does not know the exact location of the divers. Ideally, the surface vessel would know the exact location of the divers and the ability to communicate with them individually.

In recent years, there have been significant developments in the field of both analogue and digital underwater acoustic voice and data communications[1.2]. These improvements have been driven by a commercial need for underwater communication, for such application as pollution monitoring, collection of scientific data from underwater instruments, remote control of underwater vehicles, and many others. Due to the increase in the number of requirements for underwater communications, it is inevitable that it has attracted the attention of researchers. However, the technological achievements in this field have not been as spectacular as those in the wireless radio communications. This is due to the complex nature and limitations of the underwater communication channel, in which multipath reverberations can distort signals.

The recent advances in powerful Digital Signal Processor (DSP) technology and complex algorithms have enabled real-time processing applications to overcome the limitations of the channel. These technological developments, when combined with digital communication techniques, have provided transmission of high-rate data through-water.

The development of a Point-to-Point (PP) link that facilitates the transfer of data acoustically is not a simple task. Consider the case where there are multiple transmitters and receivers all sharing the same transmission medium. This is the situation when considering a medium (5 to 100m) baseline positioning system, with many divers and an array of seabed transponders, all with two-way communication capabilities.

The thesis presents a comprehensive new design methodology for underwater acoustic communications and position localisation, using a network of intelligent seabed transponders. The initial aims of the research are to design a number of digital embedded microprocessor systems, to implement networking protocols to enable the units to function autonomously, to develop a suitable packet transmission communication technique and to implement position fixing algorithms. This thesis presents the work done to develop such a system, together with the results to validate the success of the project. The main considerations in the design of the underwater navigation and communication network are that it is flexible, intelligent, and requires no user interaction during deployment and operation, hence it functions autonomously.

1.2 Organisation of the Thesis

The thesis has several themes due to the diverse nature of the research, which encompasses four main areas: underwater acoustics, position localisation, networking and communications. The emphasis of the thesis is on practical implementation, with real results supported by simulated data. The practical nature of the project highlights problems that could go unnoticed when relying purely on simulated data.

1.2.1 Chapter One: Introduction

Chapter One gives a general introduction of the topic of the thesis and introduces the reader to the basics of sound propagation in the underwater environment.

1.2.2 Chapter Two: Review of Acoustic Positioning Systems

Chapter Two provides a comprehensive review of underwater communications, underwater navigation systems and underwater acoustic networks. The emphasis is on communications systems applicable to the channel topology in which an acoustic positioning system may be used. Also, the review includes current commercial acoustic positioning systems and discusses the problems and merits of such systems.

1.2.3 Chapter Three: System Design

Chapter Three focuses on the system hardware design, which forms the platform for the research. For the positioning system to be capable of calculating the position of a mobile unit in three-dimensional space, a minimum of three-seabed transponders and one mobile unit are required. The electronic and mechanical design of a full hardware system is discussed in detail. Practical problems, such as the omni-directional projector/hydrophone bandwidth, pressure housing design, system efficiency and other limitations are also discussed.

1.2.4 Chapter Four: Network and Protocols

Chapter Four presents the protocols developed to allow the transponders to discover other transponders in range, dynamically assign identification addresses, resolve contentions and calibrate the seabed transponder array and begin positioning without user intervention. A review of several different networking protocols that were used for development ideas are also presented. The transponder control state machine and protocol layers are presented in a diagrammatic form, from which the C-programming language code was written.

1.2.5 Chapter Five: Communication Techniques

Chapter Five presents details of the need for data communications and techniques for system implementation. The communications between transponders is of paramount importance to the design of this positioning system as there is no main surface controller. In most commercial systems the surface unit controls the medium and the seabed transponders just respond to particular acoustic commands. They do not negotiate and comply with medium access rules. The communication techniques employed are designed for omni-directional transmission from a ball hydrophone. The communication technique has been designed specifically for short packet transmissions, as only a small amount of data is communicated in the encoded signal. However, the arrival time of the data packet also conveys information as to the separation distance. The coherent PSK communication technique requires the receiver to be synchronised to the carrier signal. This is achieved by transmitting a packet header, which allows the receiver to perform frequency estimation and synchronise onto the captured data. Also contained in the header is a start-bit (or start code), which is used as a timing datum, for both data decoding and time-of-flight measurements.

1.2.6 Chapter Six: Position Fixing

Chapter Six presents the time measurement accuracy of the system and discusses the problems associated with this level of accuracy. The single distance measurements are then combined to give two and three-dimensional positioning. Controlled acoustic experiments are performed in the acoustic test tank at Loughborough University to evaluate the practical attainable accuracy of the system. The position-fixing algorithms embedded in the transponder units are described and simulated, and the simulated and experimental results compared. The concept of inexpensive pingers[1] to mark survey or construction baselines is presented. The use of pingers in this manner will significantly reduce, if not eliminate the requirement to perform a geodetic calibration more than once.

---

[1] A pinger in underwater acoustic terms can be described as a device that emits an acoustic signal periodically and does not receive signals.

1.2.7 Chapter Seven: Conclusion

Chapter Seven concludes the thesis with a summary and provides a number of suggestions to improve the features and performance of the digital underwater positioning system.

1.3 Aims of the research

The aims of the research were (1) to develop a randomly deployable intelligent seabed network of transponders that autonomously assigns a Master transponder unit to control the entire system; (2) to design medium access protocols for the slow acoustic propagation environment that enables the Master transponder unit to discovery other transponders in range and assign individual identification addresses; and (3) to implement positioning algorithms on the embedded system, hence allowing the system complete operational autonomy. The system has to be capable of measuring the time-of-arrival of a communication packet to less than a few microseconds, hence offering a theoretical range measurement precision of millimetres. If the transponders are detected as stationary then they will be included in the baseline array calibration procedure. The communication technique enables the transfer of data between all transponders, which function from identical source code.

1.4 Introduction to the Underwater Environment

Sound transmission is the single most effective means of directing energy transfer over long ranges[1.3]. In turbid, saline water of the sea, both radio waves and light are attenuated to a far greater degree than sound[1.4]. Due to the relative ease of acoustic propagation in water, it is thus a topic of extreme military and commercial importance.

Considering that the globe is two-thirds water very little is known about the depth of the oceans; however, underwater acoustic experiments were conducted over five hundred years ago. In 1490 Leonardo da Vinci[1.5]: experimented by listening to ships far away using a long tube with one end placed in the water. In 1826 Colladon and Sturm[1.6] recorded the speed of sound underwater in Lake Geneva by carrying out a simple experiment in which an underwater bell was struck at the same instant as a light flashed.

By measuring the difference in time-of-arrival the speed of sound in water could be calculated.

Underwater bells were in use for navigation by the turn of the twentieth century, both as a method of detecting buoys and lightships in fog and as a method of distance measuring. In 1912, a few days after the *Titanic* disaster, a patent application was received by the British Patent Office from L.F. Richardson for echolocation with airborne sound[1.7]. A month later the same man applied for the patent for its use underwater.

The outbreak of World War 1 increased the impetus of research into the area of underwater acoustics. The three main materials used for constructing transducers for underwater applications are: (a) magnetostrictive, (b) piezoelectric and (c) electrostrictive[1.8]. Magnetostrictive transducers have been tested with limited success. The discovery of the piezoelectric effect in 1880 allowed the first hydrophones to be developed. In 1917 Langevin used piezoelectric crystals to replace the condenser versions that had been tested up until then, and in 1918 echoes were received for the first time from a submarine, occasionally at distances as great as 1500 metres. Meanwhile, a top secret system was being developed by the allies, known as ASDIC[1.8] (Allied Submarine Devices Investigation Committee). This echolocation system could determine the distance and bearing to an object, thus determining its position. In the years since then, systems have been developed not only for military work but also for civil navigation. The term ASDIC was gradually replaced in the United States by SONAR, which stands for SOund NAvigation and Ranging.

1.4.1 Physics of Sound in the sea

Physics is based upon devising a model to represent natural phenomena and making an association by measurement. Considering the underwater environment, the transport medium is the ocean - to model this, it can be represented by a three-dimensional lattice of elastically interconnected particles. The elastic interconnection between the particles allows a disturbance to propagate outwards from the location of the initial displacement. Sound is therefore, a longitudinal wave motion, which can exist in any compressible transport medium.

The rate at which a disturbance propagates though a medium is the speed of sound. A mathematical study of the physics of sound leads to the formulation of 'wave equations' which are differential equation inter-relating the spatial and temporal partial derivatives of pressure. A quantity, which determines the speed of sound, is incorporated in these equations as a 'constant of proportionality'. A further 'constant of proportionality' is a quantity relates the scale of displacement of particles in the medium to the amplitude of the pressure actually producing the displacement. This quantity is the analogue of resistance in Ohm's Law for electrical circuits. Also, the analogue of pressure in acoustics is voltage, and the analogue of particle velocity is current. The constant of proportionality relating pressure and particle velocity is known as the acoustic impedance of the medium and is solely determined by the density and elasticity of the medium - as is the speed of sound[1,3].

Young's Modulus

When considering the elasticity of a material, one often thinks of Young's modulus, which is the ratio of stress and strain within the material.

$$\text{Young's Modulus} = \text{Stress/Strain} \quad \text{(1.1)}$$

Young's Modulus is the slope of the stress-strain graph, so that for an incremental stress Δs we have a strain $\Delta L/L$, where L is the length of the bar of material and ΔL is the incremental increase in length. However, in underwater acoustics Young's Modulus, which is fine for bars of material, is inappropriate, as much of what we have to do is concerned with changes in the volume.

Bulk Modulus of Elasticity

The elasticity of a fluid is defined as the Bulk Modulus of Elasticity, K, which is still stress over strain; however, force per unit cross-sectional area is replaced with changes in pressure. Hence ΔP and linear strain becomes volumetric strain, which is the incremental change in volume ΔV for volume V. The Bulk Modulus of Elasticity, K is:

$$K = -(\Delta P/\Delta V)/V \quad \text{(1.2)}$$

The minus sign is because the volume gets smaller as pressure increases.

Shear Modulus

There is also another modulus, the Shear Modulus, which holds considerable interest for the geotechnics, seismics, civil and mechanical engineering communities. Solids sustain shearing forces; fluids do not. By contrast, both solids and fluids will sustain compressional waves. The Shear Modulus, G, is the ability a solid substance has to resist deformation by shearing. Again we have a shear force and a shear strain and the shear modulus is defined as a ratio of stress to strain. The Shear Modulus is given by equation (1.3).

$$G = \Delta s/(\Delta L/L) \qquad (1.3)$$

where the incremental shear stress is $\Delta s = \Delta F/A$, F is force and A is area.

Poisson's Ratio

Poisson's Ratio, v, describes the effects of both tensile and shear forces in operation. Poisson's ratio is a measure of how, for example a solid bar becomes thinner when pulled. For most materials, v is about 0.3, rubber is 0.5 and some materials, such as PolyTetraFluoroEthene (PTFE) has a negative Poisson's ratio. That is they get fatter when they are pulled.

$$v = \text{Diametric Strain/Longitudinal Strain} = (\Delta D/D)/(\Delta L/L) \qquad (1.4)$$

The above section is to introduce the reader to some basic physics of sound propagation.

1.4.2 Sound Velocity in Ocean Water

Sound speed, c is itself determined through the Wave Equations by two physical properties of the medium, namely its Bulk Modulus, K, and its Density, ρ. The inter-relationship between these quantities is given in equation (1.5), which is attributed to Newton.

$$c = \sqrt{\frac{K\gamma}{\rho}} \quad \text{.............} (1.5)$$

In distilled water at 20°C and at standard atmospheric pressure, the physicist measures ρ as 998 kg.m^{-3} and K as 2.18E^{9} Nm^{-2}. Thus, the sound speed is calculated to be 1481 m s^{-1}. Chapter 6 looks at practical ways of measuring the speed of sound in water acoustically, but it is often required to predict the sound speed. It should be noted that density and elasticity are quantities that are dependant on temperature, T, pressure, P, and chemical composition of the medium. The chemical composition of seawater is expressed in terms of salinity, S, or more recently in terms of electrical conductivity, G. Consequently, sound speed may be expressed as a function of temperature, pressure (depth) and salinity. There are empirical formulae to calculate the speed of sound based on the three variables (T, P, S); equation (1.6) is used later in Chapter 6 and is an example of one of them.[1.9]

$$C = 1492.9 + 3(T-10) - 6x10^{-3}(T - 10)^2 - 4x10^{-2}(T - 18)^2$$

$$+1.2(S - 35)-10^{-2}(T - 18)(S - 35) + H/61 \quad \text{.............} (1.6)$$

It can be seen that the speed of sound increases with temperature and depth; hence for an increase in depth, there are opposing tendencies. The sound velocity in the first few hundred metres of depth is complicated by diurnal changes and by mixing of the surface layer by wind and waves. A deep-sea sound velocity profile can be divided into four layers:

- Surface layer; a layer of isothermal water mixed by the action of wind on the surface of the sea. Sound tends to be trapped in this layer by surface reflections and upward refractions.
- Seasonal thermo-cline; temperature decreases with depth. During summer and autumn the thermo-cline is strong and identifiable and merges with the surface layer during the winter and spring periods.
- Main thermo-cline; seasonal changes have little effect. The main increase in temperature over that of the cold depths of the sea occurs in this layer. Although the

pressure increases with depth the net effect of temperature and pressure changes is a reduction in sound speed through this layer.

- Deep isothermal layer; constant temperature of about 4°C right to the bottom. The speed of sound increases with increasing pressure.

Only the first two layers are of concern throughout this project; however, the speed of sound is of paramount importance when designing an acoustic positioning system.

1.4.3 Sound Reflection and Refraction

Specular sound reflection obeys the same law as in geometric optics, with $\theta_1=\theta_2$.

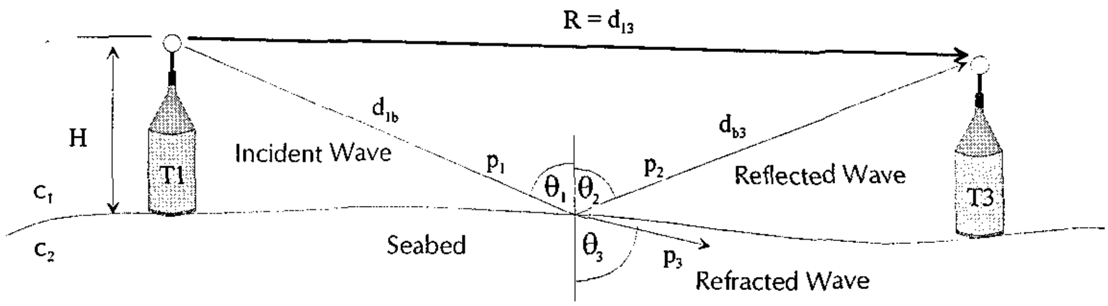

Figure 1.1 Reflection and refraction at a boundary

Sound refraction obeys Snell's law, with

$$\sin(\theta_3)/\sin(\theta_1) = c_2/c_1 \qquad (1.7)$$

Acoustic energy transfers from a lower to a higher acoustic impedance medium, e.g. sound penetrates the water at an air to water boundary, irrespective of the angle of incidence. Total internal reflection can occur at sufficiently low grazing angles if transmission is attempted from a higher to a lower acoustic impedance. This is known as the critical angle θ_c, and occurs when the angle θ_3 increases to graze along the interface, so that θ_3=90°. This marks the onset of total internal reflection; then,

$$\sin(\theta_3) = 1 \quad \text{and} \quad \theta_1=\theta_c= \sin^{-1}(c_1/c_2) \qquad (1.8)$$

The acoustic impedance of the materials on either side of a boundary, i.e. water to seabed or water to air, determines the degree of reflection or transmission across the boundary. The pressure reflection and transmission coefficients are extremely important when designing transducers, determining the seabed sediment properties, sonar modelling and in calculating target strengths.

1.4.4 Sources of noise in the ocean

Acoustic noise is of considerable importance because it affects the performance of all acoustic systems. Sources of noise include the following: seismic disturbances, oceanic turbulence, non-linear wave interaction, ship traffic, surface waves, and thermal noise. Also, there are intermittent noise sources, such as precipitation, biological sounds, explosions, seaquakes and volcanoes. Figure 1.2 shows the spectrum of noise in shallow water and the effects of wind and shipping are clearly indicated. As can be see from the plot the quietest frequency band of the ocean is around 80KHz, hence the communication frequency band used during this project.

The term ambient noise may be said to be the noise of the sea itself. The measure of ambient noise is that part of the total noise background observed with a non-directional hydrophone, which is not due to hydrophone and mounting. The noise generated by the hydrophone, mount and pre-amplifier is called "self-noise".

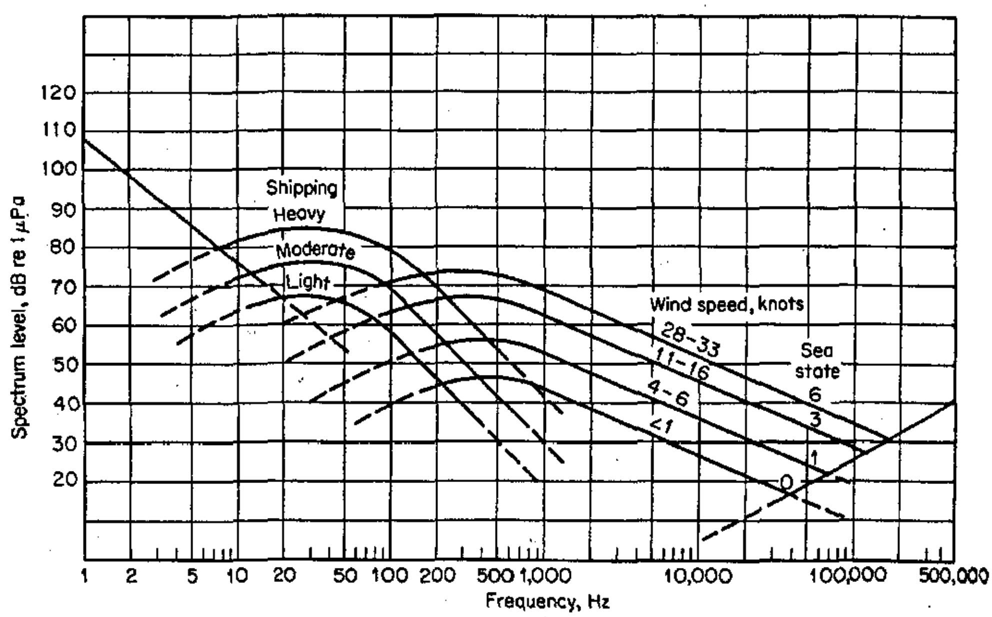

Figure 1.2 Noise in the ocean, shallow water[1.4]

The variability of ambient noise with time is clearly identifiable due to causative processes, except for thermal noise. It is, in general, considered that noise due to shipping exhibits the shortest-term variability. Wind induced noise, by contrast, exhibits statistics, which vary relatively slowly because of the inertia imparted by the time required to build up, or dissipate, a high sea-state sea. The variability of ambient noise level decreases with depth. This is because the noise is generated at the sea-surface by wind or shipping and because, as depth increases, so also does range-dependant attenuation caused by sound absorption. However, in the low frequency band (10Hz to 100Hz) where shipping noise may predominate, there is only a small decrease with depth. This is because of duct-like propagation within the deep ocean sound channel.

1.4.5 Reverberation

All acoustic systems, whether active or passive, are subject to the corrupting influences of noise. Active systems, however, give rise to another source of corruption known as reverberation[1.10] which is intimately associated with several interlinked physical effects. These effects are

- multipath propagation caused by boundary (seabed and sea-surface) reflections;
- multipath propagation caused by a possible multiplicity of refractive transmission paths, such as thermal microstructure and flow velocity microstructure;
- Surface scattering caused by sea-surface and seabed roughness or entrained air bubbles in the immediate surface layer;
- Volume scattering caused by suspended reflective and diffractive objects, such as plankton (drifting) and nekton (swimming) animal and plant life forms in the ocean.

Sonar systems including acoustic communication links are either noise limited or reverberation limited. If a system is noise limited then increasing the signal power will have the effect of improving the signal-to-noise ratio. In the main, if a system is reverberation limited, the corruption induced by volume and surface scattering will, because of a large number of scattering entities, be largely incoherent. Hence, increasing the signal power will not necessarily improve the performance of the system. The same may well be true of multiplicity of refractive transmission paths. However, reverberation caused by multiple specular reflections from the sea-surface and seabed will result in attenuated and delayed replicas of the transmitted signal being received at the receiver.

1.4.6 The SONAR Equations

The sonar equations are used to design a system with a specific performance, or to predict the performance of an existing system. Underwater acoustic systems inevitably involve the detection of signals. Whilst sonar systems can be subjected to a predominating Gaussian noise corruption, the fact that excellent reflectors bound the sea - its surface and floor - means that reverberations may in some instances present a far greater problem. The sonar equations are established by combining information relating to the source power output and directivity, transmission loss and noise or reverberation corruption. The sea is an extremely complex transmission medium and the sonar equations produce guidelines, rather than exact results.

The basic sonar equation expresses the difference between the signal-to-noise ratio at the output of the projector and the Detection Threshold (DT) at the output of the receiver. This difference is the Signal Excess (SE). In decibels (dB) the sonar equation is as follows:

$$SE = S - N - DT \quad (1.9)$$

where Signal power (S) is in the analysis bandwidth and the Noise power (N) is in a 1Hz bandwidth.

Terms are added to the basic sonar equation (1.9) for active and passive sonar systems. Passive SONAR systems detect signals radiated by a target; hence the basic passive sonar equation is therefore:

$$SE = (SL - TL) - N - DT \quad (1.10)$$

where SL is the Source Level and TL is the Transmission Loss.

Active sonar systems detect target echoes; hence they project a sound source towards the target and listen for an echo. The basic active sonar equation is therefore:

$$SE = (SL + TS - 2TL) - N - DT \qquad (1.11)$$

Where TS is the target strength of the object reflecting sound back towards the sonar system. In active sonar systems there is a two-way propagation loss, hence the transmission path is from the source to the target and back to the receiver, assuming the receiver and source are at comparable locations.

Although this thesis does not present the development of a military sonar system, the sonar equations are still used to predict the maximum likelihood of detection of a communication signal. When designing a Point-to-Point (PP) communication system the receiver sensitivity and output power are used in the sonar equation to calculate the theoretical operating range, or vice versa. The sonar equation estimates the propagation losses or transmission loss.

Transmission Loss

The intensity of a pressure wave decreases as it propagates from the source, which is of considerable importance. The parameter that describes the decrease of intensity with distance is known as the transmission loss and is denoted by TL. The TL is the sum of a spreading loss and an attenuation, the latter caused by the unavoidable frictional conversion of sound into heat during propagation.

Spreading loss is often generalised to be either spherical (free-field) or cylindrical. Free-field conditions are approximated only when all reflecting boundaries are significantly far from the source or receiver, so that no channelling of acoustic energy can occur. This is channel and frequency dependant, as at low frequencies free-field propagation can only be assumed in deep water, whereas at high frequencies, because the attenuation per unit distance rises with increasing frequency, the effect may also be evident in shallow water. The *inverse square law* is the basic loss law for spherical spreading, giving the intensity I(r) at range r, relative to intensity a 1m standard reference range, is:

$$I(r) = r^{-2}$$

or, expressed in dB

$$I(r) = -20 \log_{10}(r) \qquad (1.12)$$

If reflections from the sea-surface and the seabed result in an acoustic waveguide, free-field conditions are not relevant. Propagation may then take place with a cylindrical spreading law, for which

$$I(r) = r^{-1}$$

or, in dB

$$I(r) = -10 \log_{10}(r)$$

However, the cylindrical law tends to under-estimate the acoustic losses because some of the sound penetrates into the seabed and the surface reflections vary depending on the sea-state. A practical law is often used, which is an intermediate law, between the spherical and cylindrical laws and is define in dB as:

$$I(r) = -15 \log_{10}(r) \quad (1.13)$$

1.5 Conclusion

This chapter has outlined the aims and extent of the thesis, along with an insight into how sound propagates. The introduction to the underwater environment is only a brief one because this general area is covered in considerable detail in other literature.

1.6 References

[1.1] US Army Corps of Engineering
NAVSTAR Global Positioning System Surveying
Department of the Army, US Army Corps of Engineers, Washington DC
Manual No.: 1110-1-1003
Pages 2-3, 5-3 August 1996

[1.2] Sari H.
Underwater Acoustic Voice Communication using Digital Techniques
Ph. D. Thesis, Loughborough University
1997

[1.3] Coates R. F. W.
Underwater Acoustic Systems
1990, ISBN 0-333-42542-1

[1.4] Urick R.J
Principles of Underwater Sound
McGraw-Hill Book Company, 3rd Edition
pp. , 1983, ISBN 0-932146-62-7

[1.5] Bell T. G.
Sonar and Submarine Detection
US Navy Underwater Sound Laboratory, 1962

[1.6] Haines T. G.
Sound Underwater
Book published by David and Charles, London, 1974

[1.7] Hunt F. V.
Electroacoustics
Book published by John Wiley & Sons Inc, New York, 1954

[1.8] Tucker D. G. and Gazey B. K.
Applications of Underwater Acoustics
Applied Underwater Acoustics, Pergamon, London, 1966

[1.9] Waite A.D.
Sonar for Practising Engineers
Thomson Marconi Sonar Limited, 2nd Edition
pp., 1998, ISBN 0-9528033-1-3

[1.10] Physics of Sound in the Sea
Peninsula Publishing, California USA
Chapter 12, ISBN 0-9321-24-4

Chapter Two

Acoustic Positioning Systems

2. ACOUSTIC POSITIONING SYSTEMS

2.1 Introduction

The world of underwater position fixing is generally divided into three areas: commercial systems, military systems and academic research systems. It is very easy to find information and glossy brochures on commercial systems; however, due to competition and commercial confidentiality it is very difficult to find out exactly what technology they are using. By contrast, in academia, work is often published in journals and international conference proceedings, enabling researchers in the field to gleen ideas from current research.

Military positioning systems are often different from commercial systems because of the need to position something covertly, for example a submarine. The positioning is therefore often performed passively, by detecting the self-noise of the target of interest. The military also use active systems to position a target by generating an active 'ping' that allows a position fix to be determined by other passive systems. Military positioning systems are often monostatic, that is, they are either listening for acoustic signatures or generating an acoustic pulse and listening for echoes from targets. The systems tend not to use seabed-positioning aids (transponders), since they are not attempting to position themselves. The distinct deference between navigation position fixing systems and target positioning systems is obvious and only those for navigation will be considered in this thesis. However, there are references to passive positioning using similar algorithms and techniques as used in military systems.

This chapter present the reader with a background in the types of positioning systems and an insight into the various types of existing Underwater Acoustic Positioning Systems (UAPS).

2.2 Commercial Systems

There are several companies that specialise in UAPS, but the world-leaders in this area are Kongsberg, Simrad and Sonardyne International. The major oil and survey companies use Simrad and Sonardyne systems and are therefore generally accepted as industry standards. Survey companies are then required by the contracting organisations to use such proven acoustic positioning systems.

Acoustic positioning is frequently used in conjunction with other positioning systems to position and track a vessel, as there is often a requirement to have redundancy in not only the position information but also the systems. Therefore, acoustics is often an integral part of an overall positioning system that possibly also has a Differential GPS (DGPS) and an Inertial Navigation Systems (INS).

2.2.1 Global Positioning System (GPS)

GPS is a well-known system that enables a GPS receiver to calculate its position to within approximately 20 metres anywhere on the surface of the earth, as discussed in Chapter One. It does this by receiving signals transmitted from satellites. GPS receivers are passive, hence they only receive signals; they do not transmit. For a GPS to operate, the receiver (often 12 parallel channels) requires an unobstructed view of the sky; hence they often do not perform well within forested areas or near tall buildings. The GPS operations depend on a very accurate time reference, which is provided by atomic clocks at the U.S. Naval Observatory. Also, each GPS satellite has atomic clocks on board.

Each GPS satellite transmits data that indicates its location and the current time. All of the GPS satellites synchronise operations so that these repeating signals are transmitted at the same instant. The radio signals propagate at the speed of light and arrive at a GPS receiver at slightly different times because the slant ranges to the satellites in view are different. The distance to the GPS satellites can be determined by estimating the amount of time it takes for their signals to reach the receiver. When the receiver estimates the distance to at least four GPS satellites, it can calculate its position in three dimensions.

Ground stations continually track the satellites, which orbit the earth every 11 hours 56 minutes[2.1] and corrects the orbits so that they are circular. Figure 2.1 shows the three components that make up a GPS.

GPS and DGPS Accuracy

The accuracy of a position determined with GPS depends on the type of receiver. Most hand-held GPS units have about 20 to 100 metre accuracy[2.1]

When the system was created, timing errors were inserted into GPS transmissions to limit the accuracy of non-military GPS receivers to about 100 metres. This part of GPS operations, called Selective Availability, was eliminated in May 2000.

Other types of receivers use a method called Differential GPS (DGPS) to obtain much higher accuracy. DGPS requires an additional receiver fixed at a known location nearby. Observations made by the stationary receiver are used to correct positions recorded by the roving units, producing an accuracy better than 1 metre. Due to the vast nature of the GPS's array of satellites it can be assumed that two receivers fairly close to each other say within a few hundred kilometres, will receive signals that have travelled through virtually the same atmosphere, and so will have virtually the same errors.

The stationary reference receiver receives the same GPS signals as the roving receiver but instead of working like a normal GPS receiver it attacks the equations *backwards*. Instead of using timing signals to calculate its position, it uses its known position to calculate timings. It calculates what the travel time of the GPS signals should be, and compares it with what they actually are. The difference is an "error correction" factor. The receiver then transmits this error information to the roving receiver so it can use it to correct its measurements.

In the early days of GPS, private companies who had big projects demanding high accuracy - groups like surveyors or oil drilling operations - established reference stations. However, nowadays there are many public agencies that transmit GPS corrections using radio beacons that are already in place for radio direction finding (these usually operate at around 300 kHz).

Many new GPS receivers are being designed to accept corrections, and some are even equipped with built-in radio receivers. Also, some academic institutions are experimenting with the Internet as a way of distributing corrections.

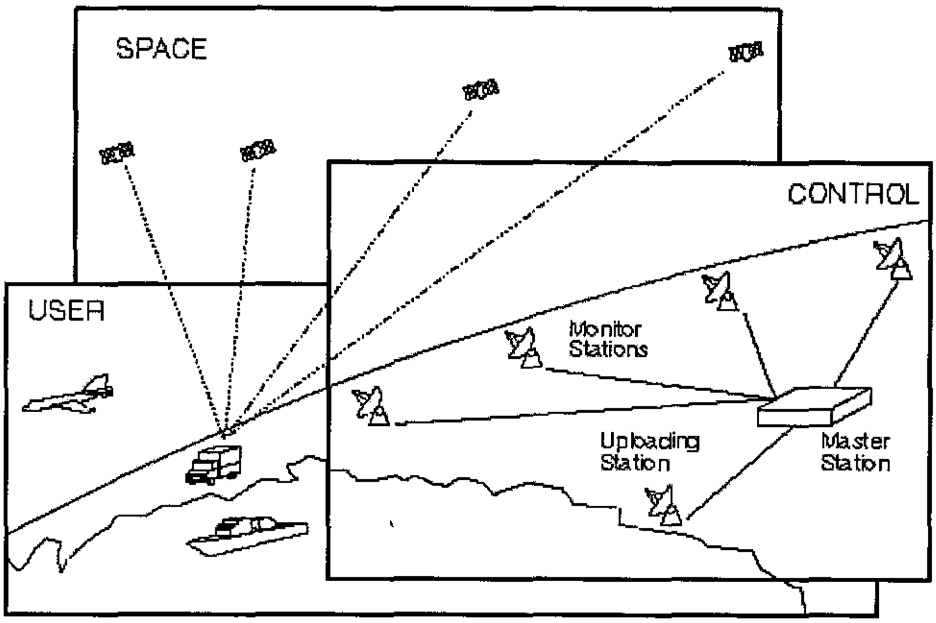

Figure 2.1 Global Positioning System component

2.2.2 Inertia Navigation System (INS)

This is an alternative method for enhancing dead reckoning. The principle of operation involves continuous sensing of minute accelerations in each of the three directional axes and integrating over time to derive velocity and position. A gyroscopically stabilised sensor platform is used to maintain consistent orientation of the three accelerometers throughout this process.

Although this method is simple in concept, the specifics of implementation are demanding. This is mainly caused by error sources that affect the stability of the gyros used to ensure correct attitude. The resulting high manufacturing and maintenance costs of this method usually make it impractical for vessel positioning system. For example, a high-quality INS, such as found in a commercial airliner, will have a typical drift of about 1850 metres per hour of operation and cost between £50,000 to £70,000. INS packages used in ground applications have shown performance of better than 0.1% of distance travelled, but cost up to £150,000. However, since the development of laser and optical fibre gyroscopes (typically costing £1,000 to £5,000), INS is becoming more suitable for short term positioning applications.

One advantage of inertial navigation is its ability to provide fast, low-latency dynamic measurements. Also, INS sensors are self-contained, non-radiating and non-jammable. The main disadvantage is that the angular rate data must be integrated once to provide orientation and the linear velocity rate data must be integrated twice to provide linear position.

2.2.3 Underwater Acoustic Positioning System (UAPS)

The underwater acoustic positioning systems produced by Simrad and Sonardyne have, in the past, been purely acoustic and are complex, requiring relatively sophisticated 'top-side' and deployment equipment to install, calibrate and interface to the complete vessel positioning system. More recently, these companies are offering a complete positioning package, thereby reducing the interface problems that frequently occur. Sonardyne, for example have a variety of acoustic positioning systems, which have been classified into the following categories:

- Long BaseLine (LBL)[2.2]
- Short BaseLine (SBL)
- Ultra-Short BaseLine (USBL)[2.3]

A popular underwater navigation technique is a long baseline (LBL) transponder system, which can have a range of several kilometres and be deployable down to depths of 7,000m. The principle of the LBL method is to determine the position of underwater mobile units relative to an array of transponders that are generally deployed in static positions on the seafloor. An interrogating acoustic module mounted on the vessel, in a tow fish or on the seabed sends out an acoustic pulse to the bottom transponders and the target-mounted transponders. Each transponder replies on its own individual reply frequency. A range-meter or signal processor measures the elapsed travel times of the acoustic waves propagating between the master acoustic module and the transponders.

After an array of transponders is deployed in a planned pattern, it can be calibrated by collecting random sets of slant ranges and processing a statistical analysis to determine the relative shape of the pattern or alternatively by measurement of baselines using the sing-around technique or bottom direct ranging.

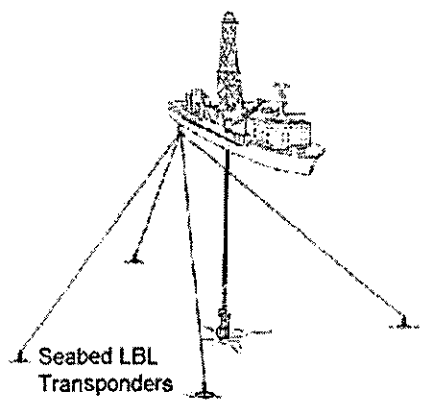

*Figure 2.2 Long BaseLine (LBL) Positioning System**

Using the separate Grid-on-Grid method by comparing simultaneously gathered sets of acoustic positions of the surface vessel given by the LBL system and an auxiliary surface navigation system completes the geodetic system calibration. Once the relative and/or geodetic calibrations are completed, the navigation of the ship equipped with the acoustic module can be accomplished within the bottom array by direct interrogation. Positioning of underwater vehicles is accomplished by means of a relay transponder and by alternating direct interrogation and relay interrogation to the bottom transponders through each target transponder. Range sets from at least three transponders are required during each cycle for use in a least-squares algorithm to determine the position of each item, with compensation for sound velocity and ray bending included in the 'top-side' software. The main advantage of the LBL positioning is the excellent and constant accuracy whatever the movements and the position of the mobile unit being tracked over the working area. The accuracy is independent of depth if environmental parameters are properly compensated. However, due to the limits of acoustic propagation, an LBL system is only practical for site-specific applications.

SBL systems generally operate in an opposite manner to LBL systems and comprise an array of hydrophones mounted at various fixed points on a vessel or rigid structure. The separation between the hydrophones is typically in the order of metres. The hydrophones are electrically connected to the same processor system that detects the arrival time of the acoustic signals.

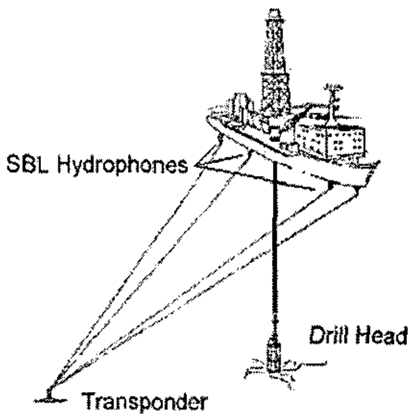

*Figure 2.3 Short BaseLine (SBL) Positioning System**

The difference in the times of arrival at the hydrophones enables a bearing to the source to be calculated. However, the system has to correct for the pitch and roll of the vessel at the instants of receiving the acoustic signal. Range to the transponder is found by measuring the acoustic propagation delay, as in LBL systems. This is an intermediate step between LBL and USBL systems.

The USBL technique uses a single acoustic array and determines the direction of arrival of the acoustic waves in both horizontal and vertical planes by phase comparison. In addition, the system measures the slant range to a sea-bed transponder by direct interrogation. As with SBL, it is essential to compensate the target transponder position for the surface vessel attitude (Heading, Pitch and Roll). The advantage of USBL is the single transducer/hydrophone array; however, noise susceptibility may cause reduction of accuracy and range limitation when using standard acoustic pulses. High signal-to-noise ratio can be obtained using a directional hydrophone, although this limits the operational coverage due to the "cone of acquisition" effect.

Techniques that can improve the efficiency and reliability of USBL systems in noisy and reverberant environments include the following: wide frequency band and coded signals; pulse compression with correlation; and simultaneous measurements of time and phase of arrival on an acoustic array with dimensions well over a wavelength.

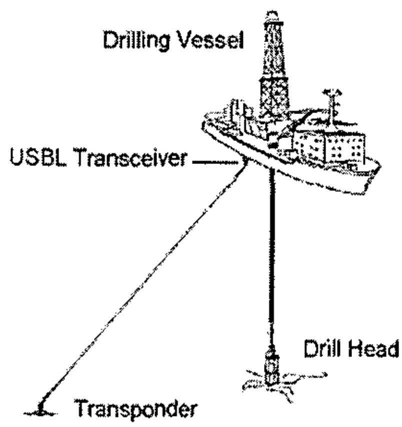

*Figure 2.4 Ultra-Short BaseLine (USBL) Positioning System**

There occasionally arises a situation when a combination of systems is required, such as LBL and USBL systems, and is termed LUSBL. The Simrad equivalent to Sonardyne's USBL is the HiPAP system. This system, as presented in the company literature, has a quoted position or range accuracy of less than 20 cm at Extremely High Frequencies (EHF 50 – 110 kHz Sonardyne's frequency band classification), hence comparable with DGPS accuracy. The UAPS frequency bands, with typical relative accuracies and operating ranges, are shown in Table 2.1 .

Frequency Bands	Relative Accuracy	Acoustic Range
Low Frequency (LF) (7.5 - 15kHz)	0.25 - 20 metres	>10km
Medium Frequency (MF) (18 - 36kHz)	0.15 – 1.0 metres	3km
High Frequency (HF) (30 – 60kHz)	0.05 – 0.5 metres	1.5km
Extremely High Frequency (EHF) (50-110kHz)	0.02 – 0.15 metres	1km
Very High Frequency (VHF) (200 – 300kHz)	0.01 – 0.10 metres	<100m

Table 2.1 Frequency bands, accuracies and operating ranges[2.5] & [2.11]

* Original diagrams courtesy of Sonardyne International Limited, UK

An important word quoted here is, 'relative accuracy'; this is the accuracy between calculated positions, hence not a reference to any particular datum. So for example; for a known position (x, y, z) of $P_1(3,4,5)$ the acoustic positioning system calculates a position of $P_{a1}(4,4,5)$, hence incorrect by 1 unit on the x-axis. However, assume that the known position moves to another known position $P_2(4,5,5)$, the acoustic position calculated will be $P_{a2}(5,5,5)$, hence the relative movement between the two co-ordinates P_{12} and P_{a1a2} is identical. So, although there is an absolute positioning error of 1 unit in the x-axis, there is actually zero relative positioning error.

The typical applications for LBL systems are:

- Marine salvage and recovery
  - Towfish or ROV positioning
  - Subsea construction and survey
  - Mineral exploration
  - Dynamic positioning of drilling vessels
  - Permanent field monitoring

As the accuracy of acoustic positioning is normally comparable or better than DGPS, the two positioning systems work hand-in-hand and often the acoustic positioning is preferred. However, due to the fact that GPS signals do not propagate any significant distance through seawater, only acoustic and inertial systems can be used on sub-sea positioning tasks.

Intelligent Transponders

The design and development of intelligent transponders was initiated to overcome the limitations of calibrating LBL arrays, since they allow direct measurements of the slant ranges to each transponder; these devices also measure and telemeter depth, temperature and salinity information so that sound velocity can be calculated[2.6]. The transponder is termed *intelligent* because it can carry out a predefined operation upon receiving the correct command, e.g. on receiving a *measure depth* command, the transponder measures the depth and telemeters the data to the surface command vessel. This can be considered to be intelligent, although it is a matter of opinion as to the actual intelligence level. To distinguish between levels of intelligent transponders, normal transponders that respond to acoustic commands in a predefined manner will be referred to as just a transponder throughout this text.

A particularly successful transponder has been the COMPuting And Telemetering Transponder (COMPATT), a microprocessor-based device for measuring the baseline distances and telemetering the data to the surface unit[2.5]&[2.7]. To measure the distance between two transponders (*A* and *B*) on the seabed, a command is transmitted from the surface control unit at a frequency f_1 that is accepted by *A*, but not *B*; this is known as transponder *A*'s Individual Interrogation Frequency (IIF). Transponder A decodes the command, starts a timer and transmits at a frequency f_2, which is *B*'s IIF. Transponder *B* decodes the command and responds at frequency f_3, which is *B*'s Individual Response Frequency (IRF). Transponder *A* receives the IRF of *B* and stops the timer; this duration is then telemetered to the surface unit, which subtracts the turn-around time of transponder *B* from the total time and divides by two to get the one-way propagation time, hence for a known sound velocity the distance can be calculated.

Another innovative system, called MARAC (MARine ACoustic) was designed to use intelligent transponders to integrate the functions of both remote positioning and general-purpose communications[2.8]. The communication technique between transponders is Frequency Shift Keying (FSK) and each transponder has a unique address code, whereas the previous system described had a unique frequency. The surface unit transmits a signal, which comprises an address code of the target transponder, an instruction code and an error detection code. Only if the transponder receives an error-free word containing the transponder's address will it reply to the surface unit.

Consider a typical, complete position fixing system that is required when attempting to position a high-tech drilling vessel during a drilling operation. Dynamic Positioning (DP) is required when maintaining the position of a surface vessel on station or above a structure on the seabed or some structure[2.9]. A DP system controls the ship's thrusters in conjunction with the UAPS, hence the latency in positional updates with a purely acoustic system can make the control algorithm intense. To reduce the position update latency a combination of positioning systems can be used, which also gives the system redundancy. An INS can calculate positions, but they will drift with time. Depending on the precision of the INS and the position accuracy required, the INS could stay within tolerance for minutes through to days. INS is often used for short term positioning because of the low latency update. Acoustic and/or DGPS position measurements can be used to correct any position error due to drift in the INS. Acoustic position updates are often relatively slow, hence a combined acoustic and INS solution is a good compromise.

When combining positioning systems, there needs to be an algorithm that decides which system is the most reliable and accurate. Kalman filters are useful for such tasks, with the inertial system generating frequent positional updates and the acoustic systems and DGPS the measurement updates[2.10]. Initially, confidence in the measurement updates has the greatest impact on the position calculation; however, with time the errors in the inertial system decrease and the confidence in the inertial position increases. This enables the system to function for periods when GPS or acoustic coverage is poor and without noticeable position information dropout. The type of system described above is used during high-tech surveying or drilling tasks and costs several hundred thousand pounds.

At the smaller diver tracking end of the market there are numerous companies, one in particular, call PLSM Instrumentation, whose system can position up to 16 divers simultaneously. The system consists of a deployable-rigid seabed hydrophone array (SBL), the 'base unit', and 'pointers' that receive signals from the base unit. The advantage of the rigid array is that the hydrophones can be connected electrically, allowing the same timing clock to be used for all the array hydrophones. The disadvantage of a rigid array is in deployment and survey area expandability. This type of system falls into the SBL positioning system category.

The systems described above all rely on measuring the two-way acoustic propagation delay, and hence with a known sound velocity the distance can be calculated. However, it is possible to acoustically position an object passively, using similar principles to GPS or synchronous systems. The problem with synchronous systems is the timing drift between units and the initial synchronisation task. However, recently Nautronix, an established Australian acoustic positioning company, announced a revolutionary system called NASNet™ (Nautronix Acoustic Subsea Network). The highly accurate long-range, multi-user acoustic positioning and navigation system takes the principles of the satellite Global Positioning System (GPS) and applies them underwater. The concept is the result of years of research and development and relies on advanced acoustic signalling techniques developed by Nautronix. The duplicated effort and costs involved in using several stand-alone systems within an offshore field is a problem the industry has traditionally had to accept, adding substantially to life-of-field costs. The NASNet™ system has been created by adapting the principles of GPS, but transformed them to operate sub-sea by taking advantage of Acoustic Digital Spread Spectrum Communication (ADSSC) techniques. NASNet™ overcomes the problems of the duplication and acoustic pollution caused by an excess of users in the same field, a frequent cause of 'downtime'. Rather than continual two-way communication, NASNet™ transmits only, so that all users can obtain their information simply by 'listening in' to the same system. The NASNet™ seabed beacons have high accuracy atomic clocks similar to GPS satellites and regularly transmit time data, that enables a unit receiving several of these information signals to calculate its position. Instead of the traditional use of several transponders, NASNet™ requires only a small number of 'stations', which sit on the sea floor in a grid. An area of $100km^2$ will require only six stations for the entire operation and, to expand the area of coverage, extra stations are self-calibrating, thus requiring significantly less time to extend the size of the field. NASNet™ can be managed remotely, so that specific applications and management requests, including checking the status of stations, can be undertaken from a vessel on the surface, 24 hours a day, or even back onshore.

This system is a very interesting recent development in the field of UAPS, but its success is still to be proven.

2.2.4 Summary of Commercial Systems

Apart from the last system discussed, all of the UAPS are based on the same principle; measurement of the two-way acoustic propagation time. Generally, all of the proven systems used for underwater position fixing and navigation calculate range by measuring the two-way propagation delay. The NASNet™ system is an exciting development in the underwater positioning environment; however there are significant differences between an acoustic 'GPS' and Radio Frequency (RF) GPS.

Most of the underwater transponders in current commercial systems are basically responders and have very little 'intelligence'. Although most of the transponders will incorporate a microprocessor for controlling how the unit responds to various commands in a known manner, the response parameter may be configurable via the top-side vessel controller unit, so generally the sea-bed transponders are 'dumb'. The transponders only respond to commands from the surface unit during calibration, hence a LBL seabed transponder array calibration can be a lengthy task.

The advantages and disadvantages of the three types of systems (LBL, SBL & USBL) are summarised below[2.11]:

The advantages of an USBL positioning systems are:

- Low system complexity
- A single transceiver at the surface
- Ship-based system, therefore no deployment problems
- Good range accuracy with time-of-flight systems

The disadvantages are:

- Minimal redundancy
- Detailed calibration of the system is required
- Absolute positioning accuracy is dependent on the ship's sensors, i.e. gyro and Vertical Reference Unit (VRU)

The advantages of SBL positioning systems are:

- Low system complexity
- Good range accuracy with time-of-flight systems
- Spatial redundancy built in
- Ship-based system
- Small transducers

The disadvantages of SBL systems are:

- System needs large baselines for accuracy in deep water (>40m)
- Very good dock/structure calibration required

- Detailed offshore calibration required
- Absolute positioning accuracy depends on additional sensors, such as ship's gyro and VRU
- Greater than three transducer deployment poles/machines needed.

The advantages of LBL positioning systems are:

- Very good position accuracy independent of water depth
- Observation redundancy
- Provides high relative position accuracy over large areas
- Small transducer
- A single deployment machine

The disadvantages of LBL systems are:

- Complex system requiring expert operators
- Large arrays of expensive equipment
- Operational time to deploy and recover seabed transponders
- Conventional systems require comprehensive calibration following deployment

2.3 Academic Research

Academic research in UAPS is relatively small, which is due to the fact that a UAPS covers several different research areas. This does not mean there are no relevant publications; it just means that the whole system has to be broken down into key subject areas. The main areas of research interest that is related to developing an UAPS are: Underwater Acoustics, Communications, Network Protocols and Positioning Algorithms.

A considerable number of papers describing research UAPS are based on describing the architecture of the system and the communication protocols between transponders. Several years ago the control of transponders was implemented in discrete logic, increasing the hardware complexity and not very flexible for research, as changes to the functionality were time consuming[2.15][2.16]. Also, in the arena of diver acoustic navigation, the system must be very compact so that they can be easily and safely operated. As most systems were logic control with a microprocessor to compute the divers position, this typical design forced the designer to use a Pulse Position Modulation (PPM) or Amplitude Modulation (AM) to convey data[2.17]. An increase in the data-rate or number of devices addressable in the system can be achieved by expanding the number of channels (frequency bands). This is simple Frequency Modulation combined with PPM, which is typically the technique employed by commercial positioning systems today.

An early diver navigation and tracking system was a continuous transmission frequency modulated sonar used like a searchlight to detect the presence of underwater obstacles that could be used as position fixing references[2.20]. Sonar has also been used with fair accuracy to track a diver by observing the echo from the diver and bubbles of exhaled air[2.21].

The advent of low-power microcontrollers has made the task of controlling and updating transponder functionality significantly easier. An example of an early microprocessor-based system is the programmable dive computer, which was reconfigured so that when used in conjunction with an intelligent fixed array of transponders it would serve as a diver positioning system[2.18]. This system calculated the position of the diver in two dimensions from the arrival times of the signal from the intelligent array of transponders.

A relatively sophisticated diver navigation system, called the Local Area Underwater Navigation System (LAUNS), was designed to locate vessels and submersible vehicles, as well as divers around a structure to depths of 600m[2.22].

Recent research interest in Autonomous Underwater Vehicles (AUV) has prompted a new look into acoustic positioning systems, including USBL systems fitted to an AUV to enable it to guide itself into an underwater docking station. The vehicle does this by first detecting pulses emitted from an acoustic beacon in the docking station, then adjusting its angle of approach so that each pulse is detected with zero phase difference on four hydrophone elements in its nose cone[2.19].

Whilst several of the systems described above have been devised for tracking divers it is often the diver who needs to know his or her own position. Systems for diver self-navigation by triangulation have therefore been devised[2.18][2.23][2.24]. Simpler hand-held acoustic range-finder system that a diver can use to measure distances to transponders; however this is strictly for position fixing rather that true navigation[2.17][2.26]. Such simple range finders have been used for locating diver shot-lines (ascent/descent line) and for re-locating acoustically marked points-of-interest (DiverTrack, Sport and Scout, Desert Star System).

A recent research study that is slightly different from the normal positioning and tracking tasks was to track the positions of dolphins around and within a pelagic trawl[2.25]. This system used a similar passive positioning system and algorithms as used by the military to position enemy vessels. A sparse array streamer was attached to the fishing net; this contained four hydrophones and pingers to enable the array to self calibrate. A fifth hydrophone was attached to the net warp (tow-line) out of the plane of the array to enable the system to find the positions of the animals in three-dimensional space. The system passively positioned the animals by capturing their echolocation 'clicks' on the five hydrophones and measuring the time-of-arrival differences. The system used to perform real-time tracking of a number of cetaceans in the vicinity of the fishing net was highly technical and in order to gain reasonable accuracy high data sampling rates and click cross-correlation was necessary.

The use of on-chip or on-board FLASH ROM enables the firmware in the processor to be upgraded without removing the Printed Circuit Board (PCB) from the housing. Such advance in technology toward system-on-chip solutions has allowed more computational power to be incorporated into seabed transponders. Such solutions have RISC architecture control processors combined with dedicated hardware for performing digital filtering, Fast Fourier Transforms (FFT), etc. The dedicated hardware is essential in a battery-powered system, as the power requirements are a fraction of what a DSP would take to perform a similar operation.

2.3.1 Position Fixing

Navigation and tracking algorithms have been considered in great detail by many authors over a period of many years, for two, three and more transponders, in terms of transformation of co-ordinates, time and range repeatability, baseline errors, slant range errors, incremental errors, multiplicative errors, transponder vertical displacement errors platform motion and so on.[2.6]&[2.19] Position fixing algorithms can be initially classified into two categories: hyperbolic and spherical methods. This chapter has only discussed spherical positioning; however, the principles of hyperbolic position fixing by surface systems such as Decca, Loran C and Omega have been applied for many years, but there have been few comparable underwater systems. The principle is to find the intersection of two or more hyperbolic lines. There are a number of algorithms available in the literature for both two-dimensional and three-dimensional position fixing.[2.9]

2.4 UAPS Overview

The UAPS developed throughout this study is to enable divers or vessels to self-navigate using the acoustic signals received. The system allows several divers to be tracked in three-dimensional space and each diver's co-ordinates to be telemetered to a remote receiver at the surface. The positions of the divers are fixed using three or more randomly deployed seabed transponders. All of the transponders are intelligent and identical. The transponders fix their own relative positions and the position of the surface or mobile transponders by an exchange of coded acoustic commands. These positions can be related to a DGPS system by performing a geodetic calibration, which is not discussed in this thesis. The divers each know their own position from a wrist-worn computer with a back-lit graphical/numerical display. As well as the current position, the display can show the tracks from the beginning of the dive, the location of the surface vessel and the instantaneous position of the other divers. Also, a transponder can be attached to a ROV or AUV to allow the position and data of the vehicle to be tracked by the surface vessel and by any divers in the vicinity. Hall-effect switches around the display allow the diver to select display modes and input information underwater via a menu-driven display. This enables the diver to mark way-points and send Short Text Messages (STM) to another diver or to the surface unit. Further advances in data transfer between the transponders and the surface can multicast the position system to an Acoustic Local Area Network (ALAN). The US Navy has been developing underwater acoustic modems that are configured for use in a Local Area Network (LAN) for various purposes, many of which require some form of wireless inter-sensor communications capability. The development of an ALAN requires sophisticated modems and protocols, with autonomous handshaking and adaptive modulation to offer both robust and high data-rate communications.[2.27]

The data encoding technique employed is Phase Shift Keying (PSK), which gives high data-rates, allowing short communication packet lengths. The communication protocols adapt to the surrounding environment and transponder positions. During the calibration period of the transponder seabed array, communication time windows are assigned to each transponder. These windows depend on the arrival of reverberations of sufficient magnitude to affect the transmitted signal arriving at the destination transponder. This allows the transponder to dynamically adjust the maximum packet length. Having short packet lengths reduces the problem of inter-symbol interfere due to reverberation and minimises the need for error correction algorithms. However, a Cyclic Redundancy Check (CRC) is encoded into the transmitted signal.

To reduce the number of multiple packet transmissions static channel equalisation is considered to allow longer packet transmissions. This channel equalising in effect removes the multipath signals that distort the transmitted signal. The baselines forming the array triangle are determined by measuring the time-of-flight of a signal transmitted from one transponder to another and back. When the initiating transponder receives the reply it can measure the time for the two-way propagation plus the response dead-time. The distance corresponding to the one-way time is calculated by subtracting the response time, dividing the result by two, and multiplying the resulting time by the velocity of sound in the water.

The minimum number of data bits to communicate the position of the diver is 48, i.e. a 16-bit number for each axis (x,y,z). With 65536 steps, and assuming a maximum operating range of ±200 metres, position resolution of 6 mm for the x and y co-ordinates is achievable. For the z co-ordinate (depth or altitude) the resolution can be far greater because the system is designed to operate at a maximum depth of 200 metres; in this case the depth resolution would be 3 mm. However, the defined maximum ranges will change automatically if the Master seabed transponder detects that ranges to particular mobile units are close to the defined maximum. This enables individual transponder to have different scaling factors on their telemetered ranges.

2.5 Conclusion

Most of the UAPS described above are complex and by no means as easy to use as the GPS, in which a GPS receiver unit, smaller than a book, can calculate its position anywhere on the surface of the earth within minutes. The NASNet™ system is an attempt to turn GPS on it head and provide a similar service to the underwater industry, but the industry is dubious as to its success.

All the systems require a set-up or calibration period before they can be used, which often requires user input to gain the best performance of the system. The communication techniques and protocols are robust, but often very slow and with the advances in underwater communications not being implemented. This may be due to the relative small gains in performance attained with increasing the acoustic data-rate, as most positioning systems transmit very little data. However, if the seabed arrays could truly be intelligent and capable of communicating data at a reasonable rate, the whole system would move towards being an ALAN, facilitating the transponders to be used for other purposes.

The system discussed in this thesis presents novel ideas to reduce the calibration period and eliminate user intervention. This autonomous positioning system has considered technologies and protocols that are used in day-to-day high-tech items, such as computers, mobile telephones, Personal Digital Assistants (PDAs), GPS, etc.

2.6 References

[2.1] US Army Corps of Engineering
NAVSTAR Global Positioning System Surveying
Department of the Army, US Army Corps of Engineers, Washington DC
Manual No.: 1110-1-1003
Page 2-1 to 2-3, August 1996

[2.2] Sonardyne International Limited, UK
Long BaseLine Navigation and Positioning System Brochure
Sonardyne, Publication Brochure
2001, website: www.sonardyne.co.uk

[2.3] Sonardyne International Limited, UK
Ultra-Short BaseLine (USBL) Brochure
Sonardyne, Publication Brochure, Issue A1
09/2001
Website: www.sonardyne.co.uk

[2.4] Kongsberg Simrad
HiPAP® High Precision Acoustic Positioning
Kongsberg Simrad, Publication Brochure
Ref. No.: (160955/A - CB055-6-20)
Website: www.kongsberg-simrad.com

[2.5] Sonardyne International Limited, UK
Mk 4 COMPATT Transponder, Data sheet types 7800/7801/7802
Sonardyne, Publication Brochure, Issue B
10/2001
Website: www.sonardyne.co.uk

[2.6] Woodward B.
Underwater navigation and tracking
Current Topics in Acoustical Research
Vol 1, pp. 323-344, 1994

[2.7] Partidge J.
Intelligent acoustic transponders, COMPATT
Institute of Electronic and Radio Engineers, Symposium on Electronic Engineering
Ocean Technology, Swansea
pp. 423 1979

[2.8] McRoberts S.
MARAC
Underwater Systems Design
pp. 23, April/May 1984

[2.9] Milne P.H.
Underwater Acoustic Positioning Systems
E. & F.N.Spon, London and New York
1983, ISBN 0-419-12100-5

[2.10] Welch G. and Bishop G.
An Introduction to the Kalman Filter
TR 95-041, UNC-Chapel Hill
pp. 1-16, 2001

[2.11] Vickery K.
Acoustic Positioning Systems; A Practical Overview of Current Systems
Journal of the IEEE, ISBN 0-7803-5190-8
Vol. pp. 5-17, 1998

[2.12] Vickery K.
Acoustic Positioning Systems; New Concepts - The Future
Journal of the IEEE, ISBN 0-7803-5190-8
pp. 103-110, 1998

[2.13] Sokratis K.K., Assimakis K.L. and Demetrios G.L.
Underwater tracking of a manoeuvring target using time delay measurements
Journal of Signal Processing Vol. 41
pp. 17-29, 1995 ISSN 0165-1684/95

[2.14] Milne P.H.
Underwater Acoustic Positioning Systems
E. & F.N.Spon, London and New York
1983, ISBN 0-419-12100-5

[2.15] Woodward B.
A Variable Code Underwater Acoustic transponder
Acoustic Letters, Vol. 6, No. 7
pp. 94-99 , 1983

[2.16] Woodward B. and Anand A.
Logic design for a self-contained diver navigation system
Ultrasonics, March 1983
pp. 73-78 , ISSN 0041-624X/83

[2.17] Woodward B.
A self-calibrating diver's rangefinder
Journal of the Acoustical Society of America
Vol. 77 (3), pp. 1000-1002
March 1985, ISSN 0001-4966/85

[2.18] Woodward B., Joyce D. A. and Niazi L.
Diver navigation with a programmable dive computer and an intelligent transponder array
Acoustics Letters, Vol. 16, No. 3
pp. 62-68, 1992

[2.19] Woodward B.
Underwater acoustic navigation and tracking techniques
Acoustics Bulletin, Vol. 25, No. 3
pp. 62-68, May/June 2000

[2.20] Colldeweih I.R., Walls E.L. and Lee R.D.
Portable sonar for frogmen
Electronics, Vol. 34
pp. 37-39, 1961

[2.21] Woodward B. and Goodson A.D.
Diver Tracking by Sonar
Acoustics Letters Vol. 13, No. 2
pp. 25-30, 1989

[2.22] Barton R.
LAUNS - A major development in underwater positioning to aid inspection and maintenance of

offshore structures.
Survey, Vol. 10
pp. 12-14, 1982

[2.23] Newborough D. and Woodward B.
Diver navigation and tracking system
Proc Oceans' 99, Seattle, Vol. 3
pp. 1581-1586, 1999

[2.24] Woodward B. and Newborough D.
Information technology advances in underwater tracking and communications
Proc UDT Pacific 2000, Sydney
pp. 134-138, 2000

[2.25] Connelly,P.R.; Woodward,B.; Goodson,A.D..
A non-intrusive tracking technique for dolphins interacting with a pelagic trawl using a sparse array of hydrophones
Proc Institute of Acoustics, Vol.19, Issue 9
pp. 193-198, 1997, ISSN 0309-8117

[2.26] Shorrock G. and Woodward B.
A multiple function sonar rangefinder for divers
Ultrasonics
pp. 16-24, 1984, ISSN 0041-624X/84

[2.27] Green M., Rice J.A. and Merriam S.
Underwater acoustic modem configured for use in a Local Area Network
Journal of Institute of Electronic and Electrical Engineers
pp. 634-638, 1998, ISBN 0-7803-5045-6/98

2.7 Companies included in the Commercial Review

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CHAPTER THREE

SYSTEM DESIGN

3. System Design

3.1 Introduction

Locating and tracking objects underwater has been an important aim of the military for many years, in particularly anti-submarine warfare. There exist commercial acoustic positioning systems that give the position of a Remotely Operated Vehicle (ROV) or diver, but they often require a large surface vessel and a team of people to deploy, calibrate and operate them. The transponders that are deployed as constituent parts of these systems are often termed 'intelligent'; however, they are generally responders that respond to a certain command signal at a particular frequency. The transponder has no knowledge of its position or the position of the other transponders. The only unit in the positioning system that has knowledge of position is the ship-based unit, which in some systems transmits the position data to the ROV via the umbilical cable or to an Autonomous Underwater Vehicle (AUV) acoustically.

Locating of an object underwater can be attained using either spherical or hyperbolic positioning. Spherical positioning systems determine position by measuring the time of flight of an acoustic signal communicated between pairs of transponders or from pairs of acoustic beacons at known locations. The receiver in the second case must have *a priori* knowledge so that the exact time of the beacon transmission can be established. Hyperbolic positioning systems determine positions by measuring the differences in the flight times of signals from the beacons. The hyperbolic receiver does not need to know when the beacons transmitted, only that they transmitted simultaneously or at known relative delays[3.1]. An active system generates an acoustic signal and waits for a response from another active device to return to its receiver. The *round-trip* time multiplied by the sound velocity in water gives the range to the object. The sound source can be at any point on a sphere[2] of radius R, where R is the range calculated from the two-way path time. With two receivers position of the source is in a plane, hence at any point on a circle radius R. For true three-dimensional positioning a minimum of four receivers is

[2] In general terms assuming an omni-directional transmitter and receiver.

required, three elements can often be used as the ambiguity can be disregarded due to the positioning or directivity of the transducers used. Multiple element transmitters and receivers are often termed arrays, which can allow the sound beam to be focused. A passive acoustic positioning system does not generate acoustic signals so it cannot determine range. Instead, the system operates in a receive mode only so it *listens* for the acoustic signal generated by the source. When the signal is received on multiple hydrophones (or transponders) the position can be computed by measuring the relative times of arrival, assuming that the co-ordinates of the hydrophones are known.[3.2]&[3.3] For a passive tracking system to work there has to be four or more hydrophones, and the positions influence the positioning accuracy. The common factor with the two above systems is that all of the elements in the array are connected together electrically; hence the time-delay due to the signal propagation along a co-axial cable can be neglected. The processing unit that calculates the position of the sound source has all the timing information and subsequent implementation of a position-fixing algorithm produces the co-ordinates. The accuracy of the computed co-ordinates is proportional to the timing precision.

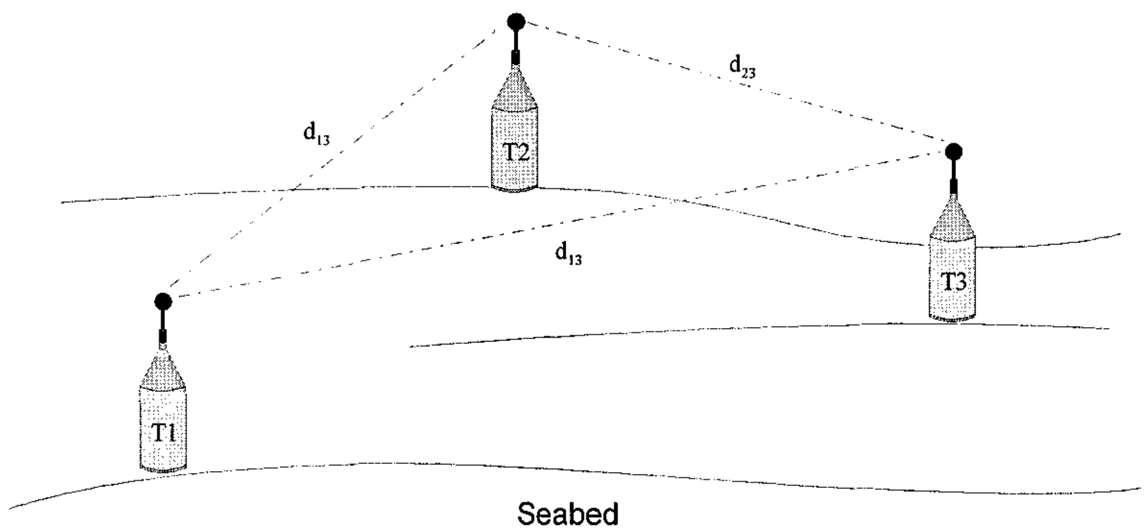


Figure 3.1 Basic transponder arrangement

Underwater acoustic navigation systems can be classified into the following categories; Short Baseline (SBL) arrays, typically less than 20m in separation, Super-Short BaseLine (SSBL) arrays, typically less than 1m and Long Baseline (LBL) arrays, typically greater than 20m separation.[3.4][3.5]&[3.6] All of the above types of underwater navigation techniques employ an array of transducers. The advantages and disadvantages of the different configurations are discussed elsewhere.[3.7]&[3.8] Using a SBL array the time measurement has to be of greater accuracy to give the same position accuracy of a LBL array due to the spatial separation of the hydrophones or transponders. Due to the difficult environment, deployment and accurate positioning of a LBL array is arduous, hence the use of smaller SBL arrays that can be deployed as rigid structures is often preferred. SBL arrays are often attached to the surface vessel; hence for position data to be referenced to the seafloor, there has to be additional equipment such as Vertical Reference Unit (VRU), a gyro and a surface navigation system (GPS).

The system described is designed to overcome the problems related to the deployment and accurate positioning of a LBL array, yet still maintain all of the advantages. The problem with a LBL array is the cabling of each array element back to a central processing unit. Precise positioning of the transponders often entails the surface vessel manoeuvring above the seabed transponder array to obtain their positions. To obviate the need for cabling or a surface vessel, each unit must be able to capture a signal and process the data, and then convey the information to another unit. Without cables the only practical way is to communicate acoustically. The positioning problem therefore depends-on a robust communication link between units and a protocol in place to prevent data collisions. The protocols and communication techniques are discussed in Chapters 4 and 5. The hardware that has been developed, and which is described here, forms the basis of the research. For the positioning system to work there has to be a system that allows the transfers of data to and from the individual transponders. The communication between transponders creates an Acoustic Local Area Network (ALAN) so in effect the transponders can be considered as acoustic modems[3.9]&[3.10]. An example configuration of the seabed transponders is shown in Figure 3.1 , where the three transponders *T1*, *T2* and *T3* are positioned randomly on the seabed. Upon deployment, each transponder *listens* for other units that may already have been deployed. If a transponder receives a communication signal from another transponder during this listening period it remains in a listening state until *discovered* by the master or primary unit. The device is now considered to be a slave or secondary unit. After a predetermined time of not receiving any acoustic data the unit undertakes the role of master transponder and begins to initiate discovery commands. The discovery sequence is similar to the protocols used in infrared communications; the similarities are discussed in Chapter 4.

When the master transponder has detected all the secondary units within its vicinity and assigned a unique address to each by an exchange of coded acoustic pulses, the next state is to calibrate the system. The calibration process is used to determine the position of each transponder relative to each other. This is achieved by measuring the time-delay between transponder *T1* transmitting to *T2* and awaiting a response from *T2*, hence spherical positioning. Figure 3.2 highlights the important timings to determine the distance between transponder *T1* and *T2*. Transponder *T1* transmits at time t_0. The time of flight of the signal is t_1-t_0, which is the information, required to measure the distance. However, *T1* does not know when *T2* received the signal as they separated by an acoustic link. Time t_2-t_1 is the duration of the data packet and is known. *T2* then introduces a set dead-time t_3-t_2, which is a standard dead-time and is known by all of the transponders. The dead-time is to prevent multipath interference from *TX1* affecting *TX2* and consequently the data packet *RX2* received at *T1*.

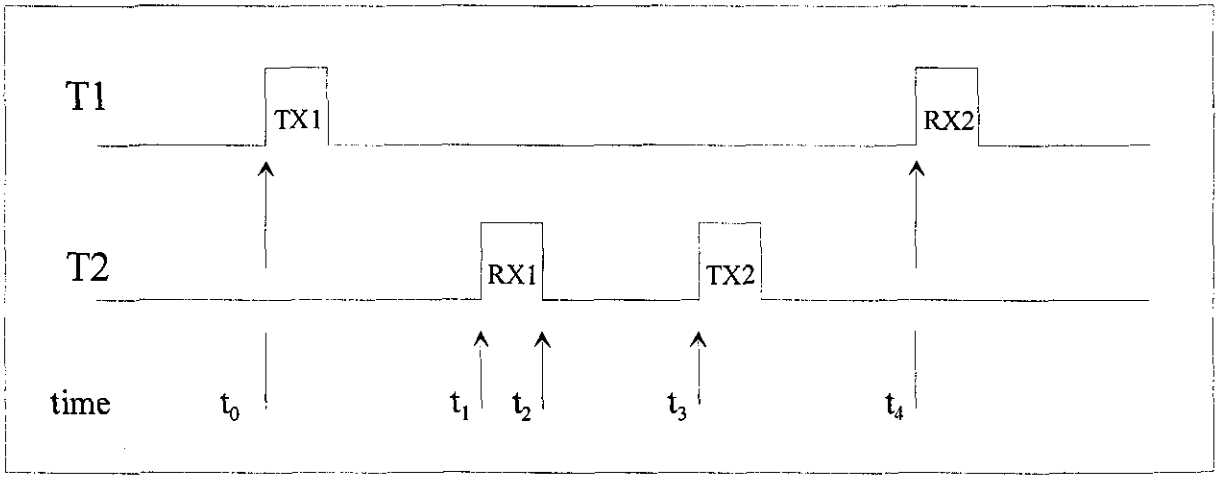


Figure 3.2 Distance measuring timing diagram

The timing information available to *T1* is the total time t_4-t_0, but t_3-t_1 is known, hence:

$$t_1 - t_0 = \frac{[(t_4 - t_0) - (t_3 - t_1)]}{2} \qquad (3.1)$$

So the distance between *T1* and *T2* is the time of flight multiplied by the velocity of sound, which as discussed early can often be assumed to be 1500m/s, so d_{12} is:

$$d_{12} = (t_1 - t_0) \times c \qquad (3.2)$$

where c is the velocity of sound in water.

Distance measurement and position fixing is addressed in more detail in Chapter 6 and will not be elaborated further here.

Each transponder has been designed exactly the same to simplify their design and deployment. The diver-worn unit has an Underwater Diver Interface (UDI) in addition, which interfaces to the main unit via the expansion port. The UDI can be connected to any of the units and allows the diver to: mark waypoints; view his position relative to the seabed transponders and surface units; view his and other diver's tracks and change the displayed data etc. The UDI has a small (128 x 64 pixel) graphical screen surrounded by six data input switches, which enables the diver to input data and change screen options underwater. The surface vessel unit is again identical to the seabed transponder, but to permit the surface observer's access to the position data it is linked to the onboard computer by a serial link (RS232). The serial link to an onboard computer enables data to be transmitted to the diver or vice versa during a dive. Also the seabed transponders could be integrated with the ship's Global Positioning System (GPS), permitting absolute position fixing. Observers at the surface can see the exact position and depth of each diver, allowing them to direct the divers to any required location.

3.2 Electronic Design

To develop such a system in analogue electronics would not be feasible, flexible or cost effective. There are a vast number of microprocessors, microcontrollers and Digital Signal Processors (DSP) available with a variety processing power. A DSP offers the processing power and special instruction to perform correlation and filtering tasks in the shortest time. However there is always a downside, which is the power it requires to perform these high-speed data signal manipulations. Considering the environment in which the transponders are designed to operate and the relatively slow velocity of acoustic propagation, there is plenty of time to decode the data transmitted, even with time-consuming signal correlation techniques employed to decipher the encoded data from the noise. The design of the system is software oriented once the basic hardware is in place, so it is advantageous to be able to program in a high level language such as C, Pascal or Basic. There are numerous compilers that will convert high-level code into assembly language for the 80x86 family. This family is the base platform of most common desktop computers (386, 486 and Pentium™ processor); hence software routines can easily be simulated on much faster user-friendly desktop machines during development. An embedded 186 microcontroller was selected upon considering the following points: processor speed, addressable memory, interface options, power supply and peripheral circuitry required. However the signal processing could be implemented using any 16-bit processor. The advantage of using an embedded microcontroller instead of a microprocessor is that it minimises the 'glue' logic. The system design can therefore be split into three distinct modules: processor module, analogue-to-digital interface module and analogue module. The circuit diagrams for the module described below are in Appendix A.

3.2.1 Processor Module

The AMD186 processor board was designed so that it could easily be interchanged and for this reason it was designed to fit in a standard PCMCIA card interface package. The high density 68-way connector gives the connectivity required for interfacing to the analogue-to-digital module. The processor module consists of the AMD186 microcontroller, 256Kbytes of Flash ROM, 512Kbytes of Static RAM and an RS232 communication port and the subsequent supervisory circuitry. A block diagram of the PCMCIA processor module is shown in Figure 3.3 . Using an embedded microprocessor reduces the number of peripheral devices required to operated as the device has Peripheral Input Output (PIOs) pins that can be configured in software to be either an input or an output pin. The boot section of the FLASH memory contains a boot program, which at power-up copies itself into the SRAM, allowing the processor the capability of re-programming the FLASH memory with a different user program. A user program can be downloaded into the FLASH memory at boot-up time if the correct command is received via the serial port. If the command is received during boot-up the processor downloads the user *hex* file via the serial link and programs the FLASH memory. If no command is received during boot-up the processor jumps to the user program and begins to execute it.

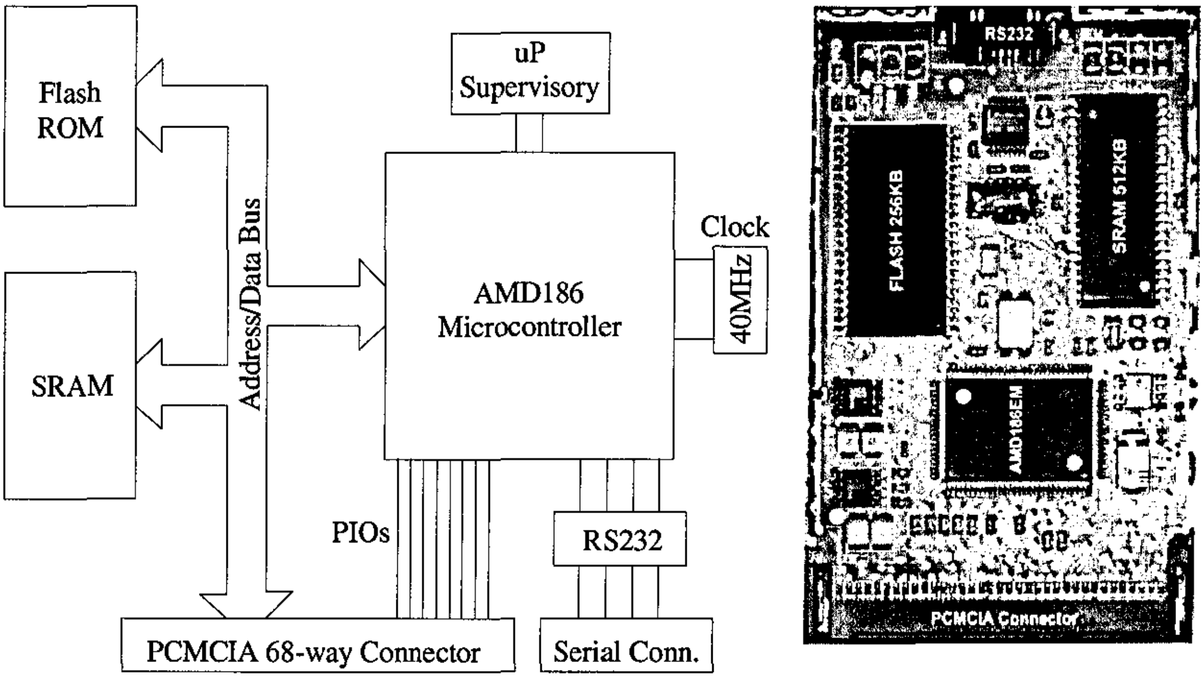


Figure 3.3 PCMCIA Processor Module

3.2.2 Analogue-to-Digital Module (ADM)

This module converts the received analogue signal to digital form via a 12-bit pipeline Analogue-to- Digital Converter (ADC), which interfaces (word aligned) directly on to the 16-bit data bus.

The transmitted waveforms are digitally generated by the processor and transferred via the data bus to an 8-bit Digital-to-Analogue Converter (DAC) that converts the synthesised digital waveform into an analogue signal. The digital data is clocked out of the 8-bit DAC at 1.25MHz; this high clock frequency permits various modulation schemes to be implemented. Other peripherals that interface to the processor module including a Real Time Clock (RTC) and a digitally controlled potentiometer that controls the input trigger level, LCD screen contrast control[3] and output signal level. The ADM can capture data at a maximum sampling rate of 5MHz, corresponding to a maximum signal capture period is 13.1ms, which is limited due to the organisation of the SRAM and the special instruction used to enable the processor to capture data at this rate. The capture rate can easily be slowed to increase the overall capture period by dividing the processor clock, i.e. a 40MHz processor clock gives 5MHz sampling, a 20MHz processor clock gives 2.5MHz sampling, and so on. A block diagram of the digital-to-analogue interface module is shown in Figure 3.4 .

[3] Only on the diver-worn unit, which has a LCD display

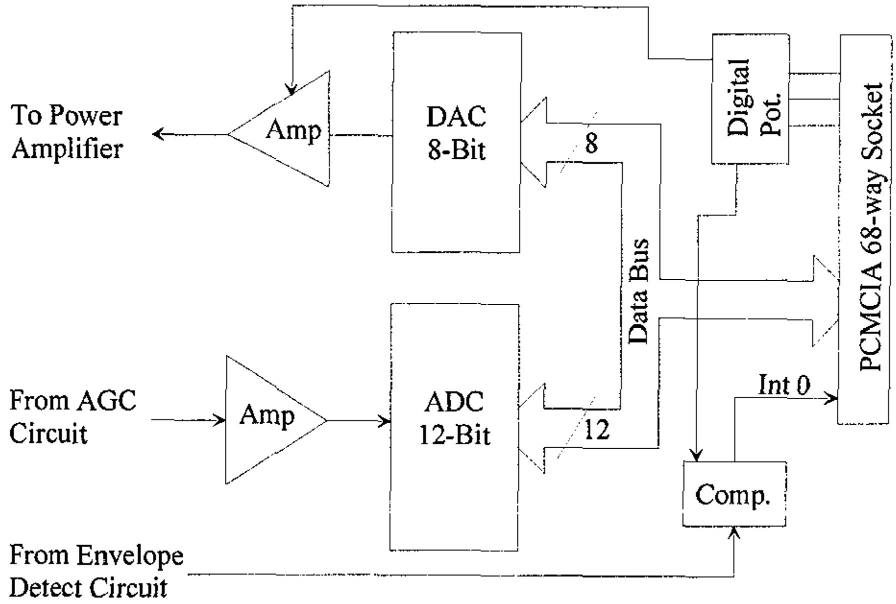


Figure 3.4 Analogue to Digital Module

Using a high sampling frequency, nearly 50 times oversampling, leads to the use of a simple anti-aliasing filter that minimises the phase distortion and when combined with digital signal processing can improve the signal-to-noise ratio. The dynamic range of the 12-bit pipeline ADC is calculated by:

$$A_{\min} = 20\log(\sqrt{1.5} \times 2^{B+1}) \quad (3.3)$$

where B is the number of bits in the ADC.

The dynamic range of a 12-bit ADC is 80dB, which is the quantisation noise level of the device. The practical dynamic range of the ADC is the point where the received signal can still be successfully decoded. For initial design purpose this was assumed to be when the signal level is below 6-bit resolution, which gives a practical dynamic range of 37dB[4]. The dynamic range of the system is important as it specifics the maximum operating distance for successful communications. The absorption of the transmitted signal over the designed operating and communication frequency is relatively small, as indicated in Chapter 1. The absorption is approximately 0.02dB/m at 80kHz, giving 4dB attenuation

[4] Communication tests show the actual practical dynamic range of the ADC in Chapter 4

over an operating distance of 200m. Assuming spherical spreading[5] (free field propagation), the Transmission Loss (TL) can be calculated as:

$$TL = 20\log(R) + \alpha R \quad (3.4)$$

where α is the absorption coefficient and R is the range.

The maximum transmission loss for a communication data packet at 80kHz occurs at the maximum specified operating range and equates to a TL of 50dB. With the 12-bit ADC having a dynamic range of 80dB a transmitted signal would be detected by the ADC. However, to increase the dynamic range of the system an Automatic Gain Control (AGC) amplifier is used. The AGC on the analogue module increases the system dynamic operating range by 42dB.

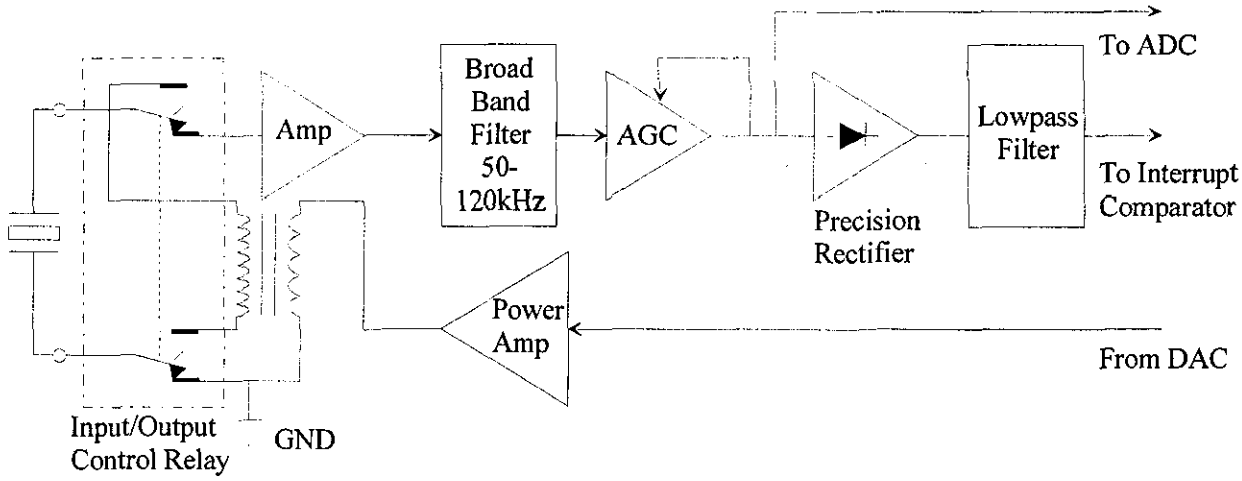


Figure 3.5 Analogue Module

3.2.3 Analogue Design

The analogue module is design to operate in two different states, which are: (1) Receive mode in which a pressure wave is converted by the hydrophone to electrical energy that is amplified and conditioned; (2) Transmit mode in which the digitally synthesised signal is amplified by the power stage to drive the hydrophone/projector, generating pressure

[5] Spherical spreading can be assumed due to the short burst data tranmission packet length, hence the received signal is via the direct path, with no constructive or destructive interference.

waves in the medium. The analogue circuitry is normally in receive-mode, hence amplifying and conditioning the signals received by the hydrophone. The received signal is conditioned by a pre-amplifier, a bandpass filter and an AGC, then fed into the ADC and an interrupt comparator. The interrupt comparator has a controllable threshold level that is under the control of the processor and can be adjusted to minimise false triggering. The use of a comparator to generate an interrupt reduces the receive-mode processing, as the processor does not have to continually *poll* the ADC to detect a signal level change.

Pre-Amplifier and Filter

The pre-amplifier is a low-noise operational amplifier (OP-AMP) that is specifically designed for transducer signal amplification. The amplified signal is then conditioned by a bandpass 8^{th} order filter to attenuate low frequency noise generated by precipitation, shipping, bubble noise and high frequency noise due to thermal agitation[3.11][3.12]&[3.13]. The characteristics of the pre-amplifier and filter are shown in Figure 3.6

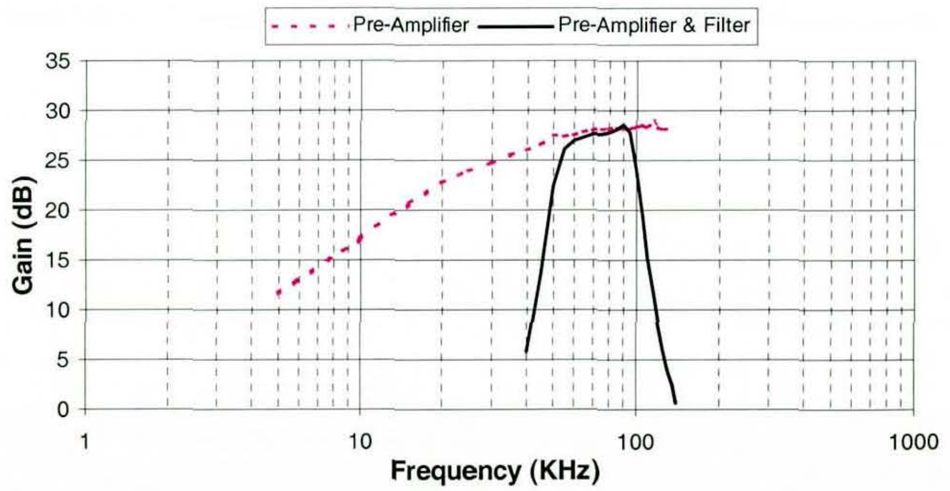


Figure 3.6 Pre-Amplifier and Filter Response

Automatic Gain Control (AGC)

The AGC circuit is designed so that the full dynamic range of the ADC is used independent of the separation of the communicating devices. The AGC amplifies the received signal by its maximum gain of 42dB until the output of the amplifier starts to exceed the prescribed output level. Then the amplification is reduced to hold the output level constant. A plot of Input Voltage vs. Output Voltage is shown in Figure 3.7 .

The output level of the AGC can be set using a potentiometer and is adjusted to below the maximum input amplitude of the ADC. The plot shows that the output of the AGC is held at the preset level over a dynamic input range of 10mV to 1V, hence 40dB.

A problem with an AGC is that it takes a finite time to adjust to the input signal and due to the design of the communication technique the data arrives in packets and is not continuous. When no signal is received the gain of the AGC is maximum, so when a data packet arrives the leading edge of the packet is over-amplified, causing clipping of the signal. If the time constant of the AGC is reduced to increase the response time of the circuit the control feedback voltage starts to ripple creating amplitude fluctuations in the data signal. To prevent this the time constant of the circuit has been designed to be asymmetric, giving a 'fast attack, slow decay', increasing the response time to an incoming packet. Figure 3.8 & 1.9 show the response of the AGC circuit to a received data packet. From the zoomed plot it can be seen that the attack time is approximately 120 times faster than the decay time. Within two cycles of the received waveform the AGC circuit stabilises to the desired output level. The transient amplitude occurs at the start of the data packet, which corresponds to the synchronisation section and has no effect on the transmitted data and due to the time latency in servicing an interrupt and beginning signal capture part of the transient is missed.

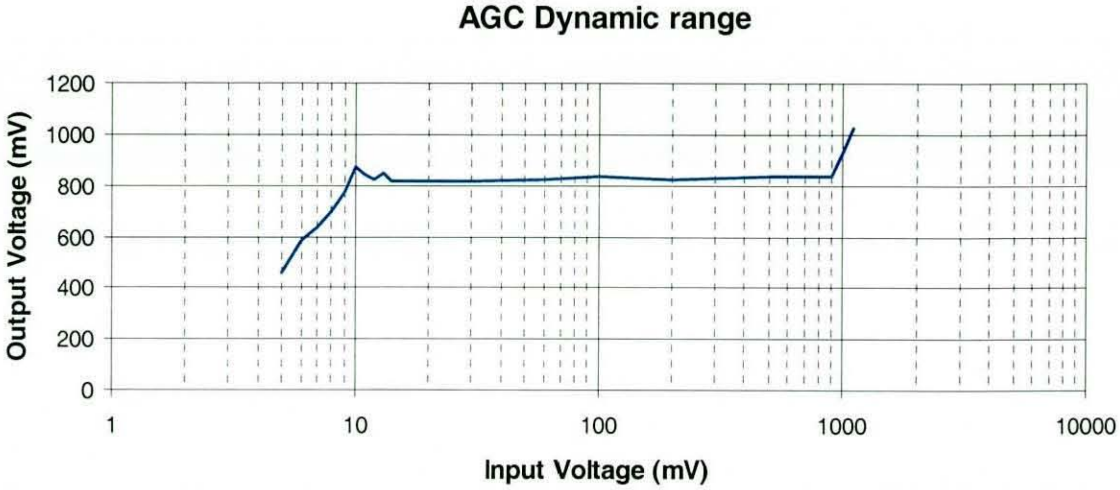


Figure 3.7 AGC Circuit Response

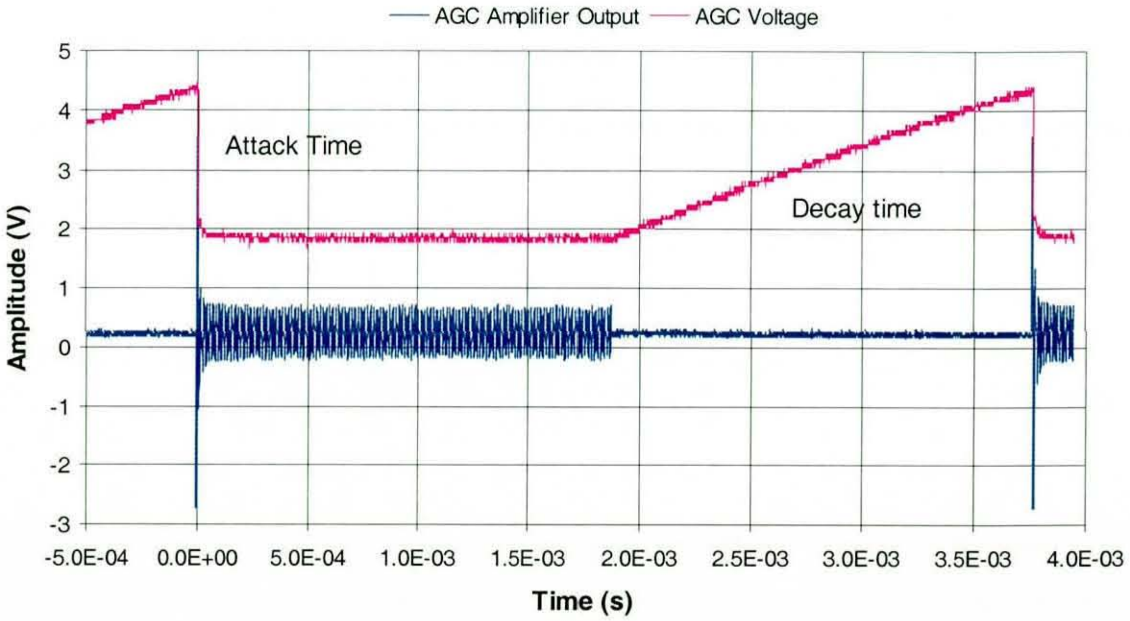


Figure 3.8 AGC amplifier response to a received signal

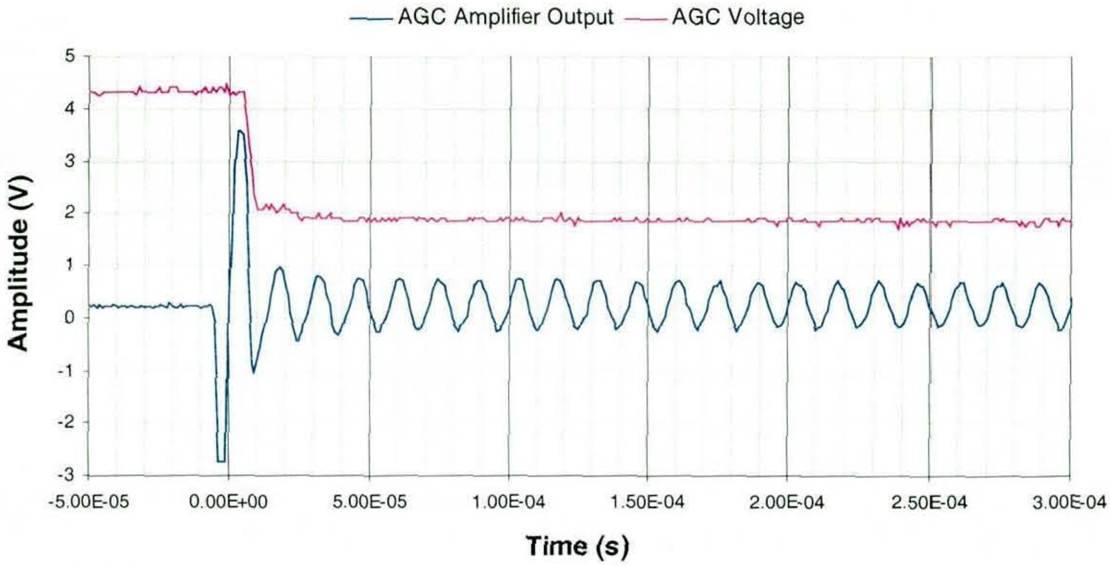


Figure 3.9 Zoomed view of the AGC Response

3.2.4 Hydrophone

In underwater communication systems the hydrophone and projector are often directional, either as a focused array or a single element. This improves the system performance as multipath signals not occurring within the receiver solid angle are greatly attenuated. The use of directional projectors and receivers in a point-to-point communication link is advantageous. However, in a three-dimensional positioning system such directional projectors and receivers cannot be used as the diver could be positioned anywhere within the operating range. Spherical 'ball' hydrophones have no directivity, hence are omni-directional. The choice of hydrophone was constrained by the following factors; omni-directionality; capable of being used as a projector as well as a hydrophone without degrading the receive characteristics; sensitivity in the frequency band of interest, which is 50kHz - 100kHz. Using a high frequency increases the power requirements for the prescribed practical operating range. Operating at a lower frequency would increase the time per bit of data, increasing the overall data packet length and subsequently reducing the data transfer rate.

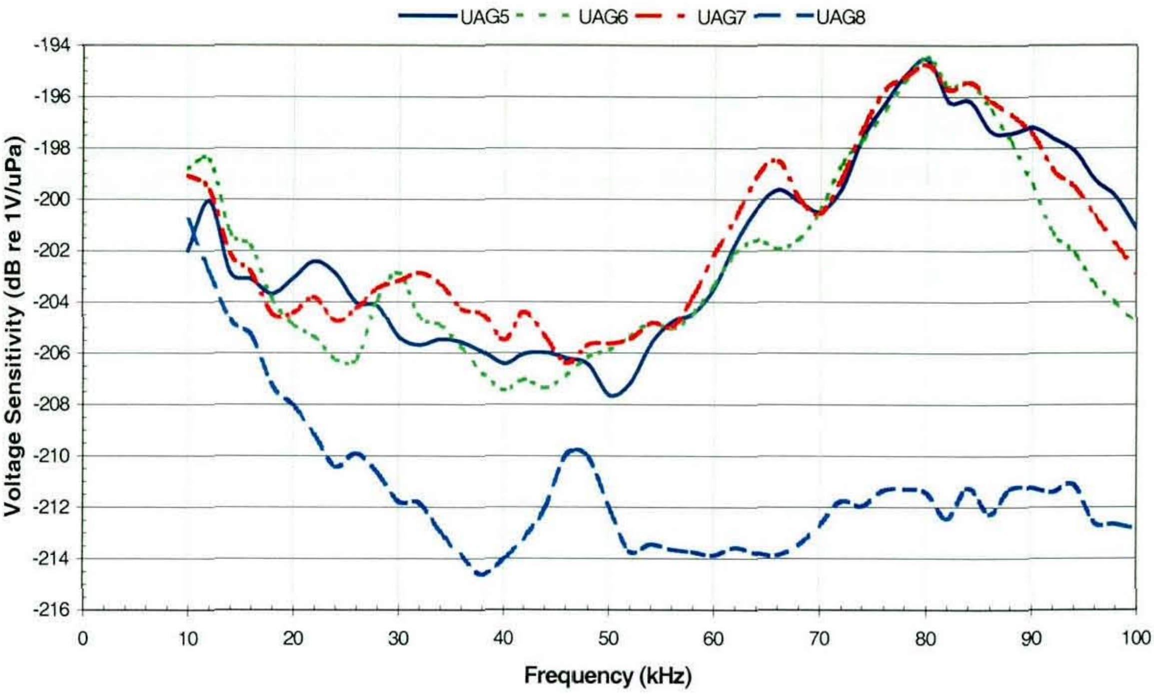


Figure 3.10 Receive sensitivity of hydrophones

The receive characteristics of the ball hydrophones are shown in Figure 3.10 where UAG 5, UAG 6 and UAG 7 are one-inch (25.4mm) diameter spheres and UAG 8 is a half-inch (12.7mm) diameter sphere. All of the hydrophones were calibrated in the Underwater Acoustic Group's acoustic test tank at Loughborough University. The calibration of each hydrophone was conducted from 10kHz (due to the physical size of the tank and the characteristics of the Brael & Kjaer reference hydrophone) to 100kHz. The calibration plots of the one-inch ball hydrophones shows that all three are resonant at 80kHz and have an approximate bandwidth of 12kHz. The quality factor Q of the hydrophone is calculated as[3.15]:

$$Q = \frac{f_0}{BW} = \frac{80}{12} = 6.6667 \qquad (3.5)$$

The quality factor Q of the hydrophone influences which communication techniques can be implemented, as the hydrophone tends to oscillate at its natural resonance and resist change. The output of a hydrophone (peizo-electric transducer) takes Q cycles to reach 0.9 of its steady state energy dissipation [3.16]. The effects and problems of using a high Q projector and receiver are discussed in Chapter 5. A calibration of a half-inch ball hydrophone has been included to show the reduced receive sensitivity across the frequency band of interest. The resonance of the half-inch ball is not shown in Figure 3.10 , as it occurs around 140kHz. To operate a system with this type of hydrophone the output power would have to be increased considerably to account for the reduced receiver sensitivity and transmit characteristics.

3.2.5 Power Amplifier

As the transponder units are battery powered, hence operating at a relatively low supply voltage, a power amplifier and transformer configuration is required to increase the output power to an acceptable level. As discussed earlier, the maximum operating range of the system is about 200m, giving a transmission loss at the extremities of operation of <50dB. Assuming a noisy environment where the ambient noise level is around 100dB, the Source Level (SL) of the communicating signal has to be greater than 150dB to be detected above the noise.

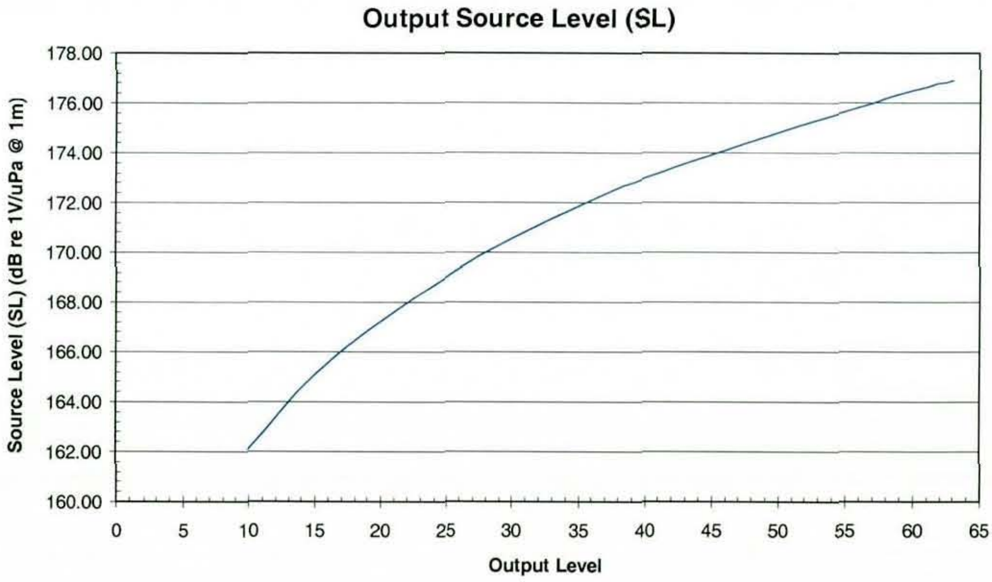


Figure 3.11 Digital output level (0-63) against Source Level (SL)

The output power of the unit is under the control of the microcontroller so that the output amplitude can be changed depending on the range to the units being communicated to when positions are known. Figure 3.11 shows the output SL range of the power amplifier module. The output levels shown on the x-axis are increments on an electronically programmable potentiometer, which is controlled by the microcontroller.

The power amplifier design is a push-pull matched FET arrangement that drives into a step-up transformer[3.14]. Normally when designing an output amplifier transformer the inductive reactance of the secondary winding of the transformer is made equal to the capacitive reactance of the projector element to achieve maximum power transfer at the frequency of interest. Figure 3.12 shows the output transformer and the hydrophone equivalent circuit.

$$2\pi f L_s = \frac{1}{2\pi f C_o} \qquad (3.6)$$

For maximum power transfer equation (3.6) is true, but when communicating over a wide frequency range there is a trade-off between output power and bandwidth. The main problems with implementing a Phase Shift Keying (PSK) communication technique is the Q of the projector and receive hydrophone, as it limits the degree of phase change per cycle with respect to the output or received level[3.16]. As discussed earlier, the bandwidth of the ball hydrophone is relatively small and implementing 180° instantaneous phase changes will cause the signal amplitude to drop significantly. The period for the amplitude to recover from the phase change is Q-cycles[3.16].

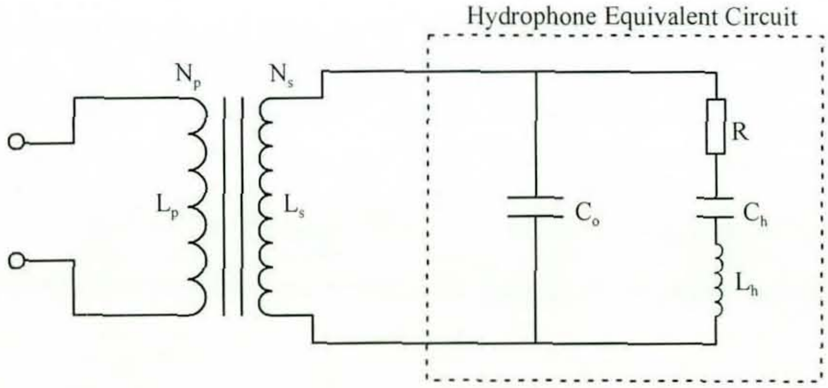


Figure 3.12 Hydrophone equivalent circuit and transformer

The secondary winding inductance cannot be used to tune the static capacitance of the transducer. The transformer is to provide a high-efficient step-up. It is necessary to recall that the high coupling coefficient of an un-gapped ferrite core results in extremely low leakage flux and thus only small values of primary and secondary leakage inductance *Lp* and *Ls*. The leakage inductance is not the same as the primary and secondary winding inductance, as this should be large. Any secondary tuning must be derived from the leakage inductance, referred to the secondary, where it appears in series with the amplifier output impedance. The referred leakage tuning inductance, L_t is:

$$L_t = L_s + N^2 L_p \qquad (3.7)$$

The Source Level (SL) of the output module at the various output levels was calibrated in the acoustic test tank at Loughborough University. Figure 3.11 shows the controllable output levels and the corresponding output power, the maximum output power being 177dB. Assume an ambient noise level as previously of 100dB the SL corresponds to a maximum operating range of greater than 3km.

3.3 Diver Interface

The underwater diver interface consists of a backlit graphical LCD with Hall effect switches around the periphery. The switches are activated by a magnet in the fingertip of the divers glove and enable the diver to input data and change display modes underwater. The six Hall effect switches perform different operation depending on the active soft keys displayed on the screen. Figure 3.13 shows the Underwater Diver Interface (UDI), with the Hall effect switches clearly labelled. Switches 5 and 6 are normally used as up/down scroll keys, e.g. used to increase and decrease the display contrast in set-up mode or scrolling through display options in normal mode. Switches 1 to 4 are soft keys and their functionality depends on the current display.

Figure 3.13 Underwater Diver Interface

Other types of UDI were considered; including a head-up display, in which information is projected onto the inside of the diver's mask similar to the technique used in military aircrafts. This type of display has been considered for Navy divers, however the technology is expensive [3.17].

Prior to using Hall effect switches reflective optical InfraRed (IR) switches were considered and tested. This design enabled a diver to operate the switches underwater without a special glove with a magnet incorporated. However, the design had several problems; higher power operation (six IR LED pulsed periodically); activated by torchlight, activation range with a neoprene glove is 1-2mm, the range is significantly increased if the surface is reflective; housing problems. The Hall effect switches operate at about 10mm range; hence they can be mounted on the inside of the housing.

3.4 Mechanical Design

Deploying electronics underwater is always problematic as the electronics circuits can easily be destroyed by a leak in the pressure housing. The design of the pressure housing has to consider the depth of operation, serviceability and interface options. The designed operating depth in the case of a diver navigation system can be considered to be the maximum dive depth of the diver, which is about 100m[6]. However, considering the other cases in which the system is a navigation aid for an ROV or AUV this depth could be increased. For this research the pressure housing was design in accordance with the expected test-site and typical dive operations; so a maximum depth of 50m was consider sufficient. Often when deploying equipment underwater or in environments where electronic circuits are likely to become exposed to the elements it is advantageous to encapsulate the electronics in a polyurethane compound. This process is often term 'potting', in the event of the housing flooding the electronic circuits will not be damaged due to exposure to sea water. This approach is ideal when a design has been finalised, however during research and development potting of expensive prototype circuitry is not advisable as it is difficult to implement changes to a circuit once potted, even with re-enterable compounds.

The water pressure on the housing at 50m is 6atm, hence $\approx$88psi. The housing was machined from PVC tube and bar to the design shown in Figure 3.15 . The unscrewable end-cap allows easy access to the battery compartment and electronics by removing an extra security bulkhead that is designed to protect the electronics in the event of a housing breach.

[6] The maximum dive depth of a diver breathing mixed gases.

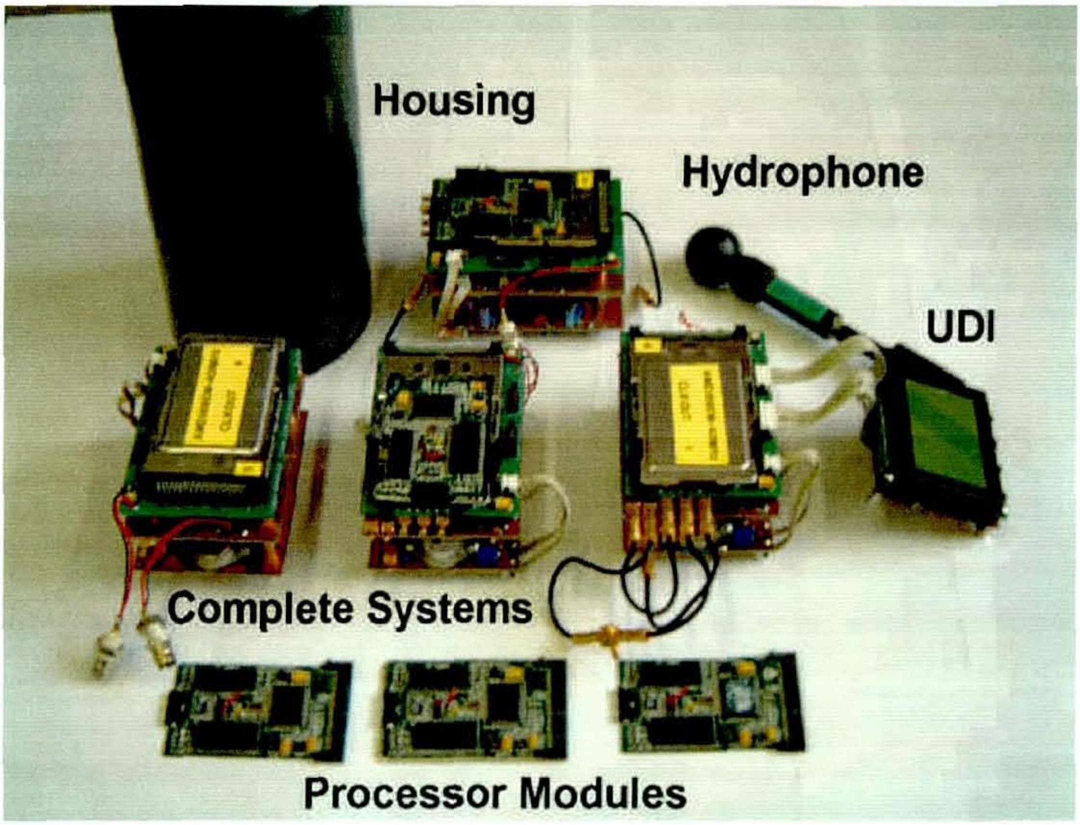


Figure 3.14 Photograph showing the various system modules

The ball hydrophone is plugged onto the housing via an eight-way Mirco-Subconn™ connector; during normal operation only two of the eight-way connector are used. The other connections are used for: recharging the batteries; reprogramming and downloading recorded data without having to open the housing, reducing the risk of leaks. A pressure sensor and status indicator is encapsulated at the top of the conical section, again to reduce the risk of leaks.

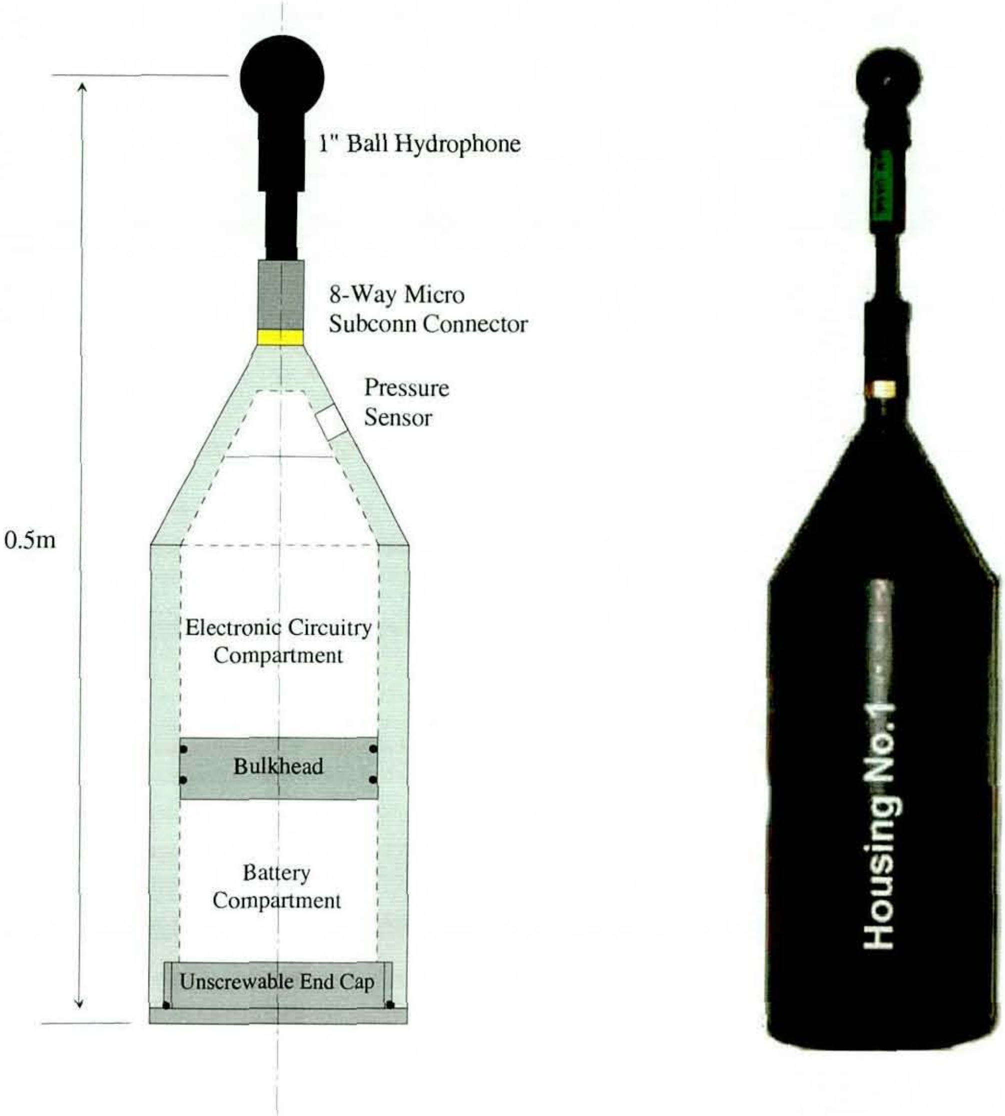


Figure 3.15 Mechanical housing design

3.5 Acoustic Beacons

Acoustic beacon are devices that only transmit acoustic signals, they do not receive signals. The design and development of efficient, low-cost acoustic beacons to mark underwater object is of significant importance in a number of areas. The beacons discussed below were initially designed to delineate the extent of a fishing net acoustically. The acoustic marking of the fishing net is not for the fisherman's but for cetaceans.

Small-toothed whales, dolphins and porpoises are all believed to possess excellent hearing and a sophisticated active sonar sense, which they employ while detecting and intercepting their prey. The problems of perceiving a fishing net appear twofold, requiring detection and classification. In general gill-nets comprise a vertical mesh curtain constructed of mono or multi-filament twine knotted together to form mesh apertures of a size that will entrap fish by their gills, or by wedging. Most polymer materials used in the manufacture of both ropes and mesh are acoustically quite closely matched to seawater and therefore relatively transparent to sound. Also, the twine is rarely more than 0.8mm in diameter and the shortest exploitable wavelength in the dolphin's sonar is around 12m. Such sub-wavelength targets intercept only a very small part of the transmitted energy. These small dimensions generate a weak non-directional echo.

To acoustically illuminate the fishing net hazard and alert an approaching animal acoustic beacons can be attached that transmit periodically. Although the original design involved considerable trials and research in to the acoustic signals and the behaviour of the target animals, the hardware platform can be used for other acoustic marking purposes. The design and development of the acoustic alarm for fishing nets is described in detail in reference [3.18]. The beacon mode *pinger* has been patented by Loughborough University and the name used 'Porpoise Incidental Catch Eliminator' (PICE™) has been trademarked.[3.19][3.20][3.21]

Due to the micro-controller design, the functionality of the pinger can easily changed by altering the firmware for different tasks. The device has been used in the following application: diver shot-rope marker, reference source, long-term equipment marking.

A new application for the pinger is to use it for marking a baseline so that the expensive transponder units can be removed from a survey field and then randomly re-deployed in the vicinity of the previous survey. The acoustic marker pinger will enable the re-deployed array to realigned itself with the previous array so that the same data from the previous survey can be used without the need for an non-acoustic reference, such as GPS and performing a geodetic calibration.

Often for small dive re-aligning the array is not feasible and post-processing the data is the only way to re-align survey point following a re-deployment of the transponder array. How the pinger can be used to eliminate the need for geodetic calibrations is described in Chapter 6.

3.6 Conclusion

The generic hardware design provides a platform for research into the various areas: network protocols, packet transmission, modulation schemes and position fixing algorithms. The modular design of the circuits and layout of the PCBs enables changes to be made to individual modules without having to change the whole design. So that various communication techniques can be experimented the hardware does not provide an optimum solution for a particular modulation, however once a modulation scheme has been evaluated and considered the most suitable a new optimised receiver, power amplifier or capture module can be designed.

3.7 References

[3.1] Deffenbaugh M., Bellingham J.G. and Schmidt H.
The Relationship between Spherical and Hyperbolic Positioning
Journal of the IEEE, Oceans' 96 Vol. 2.
pp 590-595, 1996, ISBN 0-7803-3519-8

[3.2] Freitag L.E. and Tyack P.L.
Passive Acoustic Localisation of the Atlantic Bottlenose Dolphin using Whistles and Echolocation Clicks.
Journal Acoustical Society of America, Vol. 93, Iss. 4.
pp. 2197-2205, 1993

[3.3] Moose R.L.
Passive Range Estimation of an Underwater Target
IEEE Transactions on Acoustics, speech and signal processing
Vol. ASSP-35, Iss.3.
pp. 274-285

[3.4] Milne P.H.
Underwater Acoustic Positioning Systems
E. & F.N.Spon, London and New York
1983, ISBN 0-419-12100-5.

[3.5] Woodward B.
A Variable Code Underwater Acoustic Transponder
Acoustic Letters, Vol. 6, No. 7
pp. 94-99 , 1983.

[3.6] Woodward B.
Underwater Acoustic Navigation and Tracking Techniques
Acoustics Bulletin, Vol. 25, No. 3
pp. 5-11, May/June 2000.

[3.7] Vickery K
Acoustic Positioning Systems; A Practical Overview of Current Systems
Journal of the IEEE, ISBN 0-7803-5190-8
pp. 5-17, 1998.

[3.8] Vickery K.
Acoustic Positioning Systems; New Concepts - The Future
Journal of the IEEE, ISBN 0-7803-5190-8
pp. 103-110, 1998.

[3.9] Zvonar Z., Brady D. and Catipovic J.A.
An Adaptive Linear Multiuser Receiver for Deep Water Acoustic Local Area Networks.
Journal of the IEEE, Vol. 2, pp. 389-392, 2000

[3.10] Green M., Rice J.A. and Merriam S.
Underwater Acoustic Modem Configured for use in a Local Area Network
Journal of the IEEE
pp. 634-638, 1998

[3.11] Urick R.J.
Principles of Underwater Sound
3rd Edition, Peninsula Publishing, California, USA
pp. 202-233, Chapter: The Noise Background of the Sea

ISBN 0-932146-62-7

[3.12] Mole L. A., Hunter J. L. and Davenport J. M.
Scattering of Sound by Air Bubbles in Water
The Journal of the Acoustical Society of America, Vol. 52, Number 3 (Part 2)
pp. 837-841, 1972

[3.13] Gragg R. F. and Wurmser D.
Pseudo-Doppler Resonance Phenomena in Continuous Wave Scattering from Evolving Intermediate Bubble Plumes.
The Journal of the Acoustical Society of America, Vol. 98, Number 1.
pp. 473-483. July 1995

[3.14] Lander C. W.
Power Electronics
McGraw Hill, 3rd Edition
ISBN 0-07-707714-8

[3.15] Horowitz P. and Hill W.
The Art of Electronics
Cambridge University Press, 2nd Edition
ISBN 0-521-37095-7

[3.16] Woodward B. and Chandra R. C.
Underwater Acoustic Measurements on Polyvinylidene Fluoride Transducers
The Journal of Electrocomponent Science and Technology, Vol. 5,
pp 149-157, 1978

[3.17] Gallagher D. G.
Development of Miniature, Head-Mounted, Virtual Image Displays for Navy Divers
US Army Corps of Engineering
pp 1098-1104

[3.18] Newborough D., Goodson A.D. and Woodward B.
An Acoustic Beacon to Reduce the By-catch of Cetaceans in Fishing Nets
Journal of the Society for Underwater Technology, Vol. 23, No. 3
pp. 105-114, Summer 2000, ISSN 0141-0814

[3.19] Newborough, D., Goodson, A.D. and Woodward, B.
By-Catch Reduction Acoustic Device
International Patent Classification Application No PCT/GB97/01976, AOIK 79/02
International Publication No WO 98/03062, 1998, 1p.

[3.20] Goodson, A.D., Newborough, D. and Woodward, B.
By-Catch Reduction Acoustic Device
United States Patent No.: US 6,170,436, 9th January 2001.

[3.21] Newborough, D., Goodson, A.D. and Woodward, B.
Trademark application "PICE", Porpoise Incidental Catch Eliminator.

CHAPTER FOUR

NETWORK AND PROTOCOLS

4. NETWORK AND PROTOCOLS

4.1 Introduction

Although computers often operate in a stand-alone mode there is often a need to inter-work and exchange data with other computers. It is quite obvious that the system described in Chapter 3 cannot operate in a stand-alone condition, as it requires information from the other transponder units. The design of the system requires all the transponder units to be able to communicate with each other, hence the need for a communication protocol. The way in which the data is encoded and transmitted is discussed in Chapter 5. This chapter is concerned with the network protocol, data/command packet structure and the functionality of the positioning system.

The fundamental requirements in all applications that involve two or more computers is the provision of a suitable data communications facility. However, there is a wide range of different types of communications facility that may be utilised, each intended for a specific application domain. In applications, such as transferring a file between computers within the same room or office, the communication facility will be simpler than if the file is to be transferred between computers at different geographic locations.

Irrespective of the type of data communications facility being used, in most applications data is transmitted between computers in a bit-serial mode.[4.1] Consequently, since data is transferred between sub-systems within a computer in a word-parallel mode, it is necessary to perform a parallel-to-serial conversion at the computer interface prior to outputting data, and the reverse function on the input. In addition, once data is transmitted outside a computer, there is an increased probability that the data will become corrupted. In most applications, therefore, it is necessary to incorporate not only a way of detecting an error, but also a way of requesting another copy of the affected data. This is known as error control and is just one issue that must be considered when transmitting data between computers.

Figure 4.1 shows a typical positioning system configuration; all the units are connected to a communication network, which in this case the medium is water rather than cables.

The network currently has no network controller; hence any unit can transmit at any time, increasing the probability of a data packet collision.

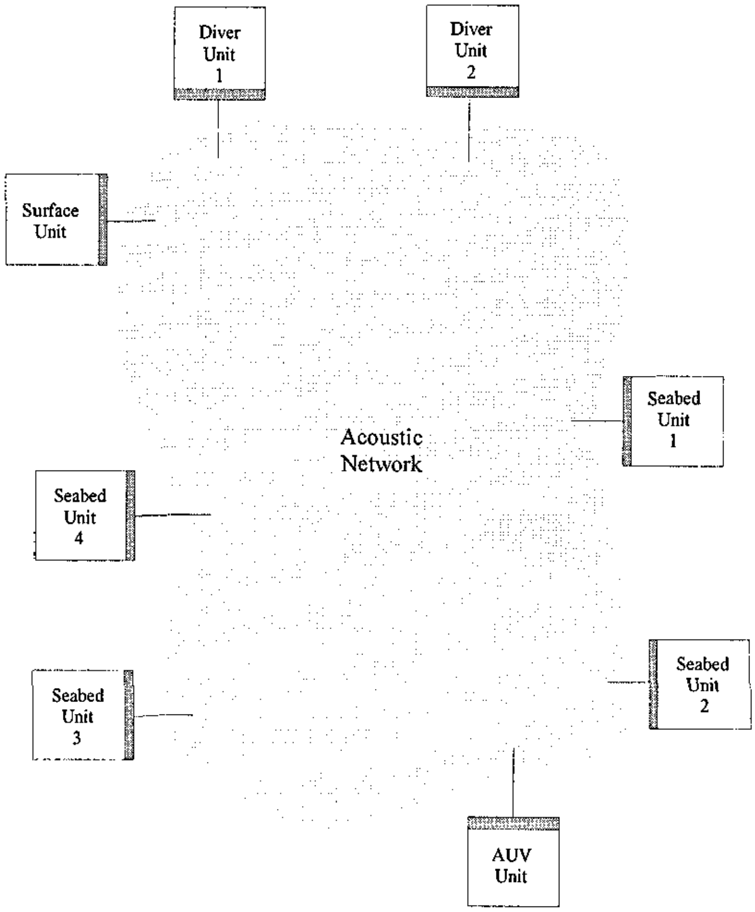


Figure 4.1 Acoustic Local Area Network

There are various network topologies; four common topologies are ring, star, hub and bus. The preferred topologies for Local Area Networks (LAN) designed to function as data communication networks for the interconnection of local computer-based equipment are bus and ring. Bus networks are normally extended into an interconnected set of buses that resembles an unrooted tree. With bus topology the single network cable is rooted around an office block or building and where a connection to the network is required a physical connection is made to the cable which is often referred to as a tap. The Data Terminal Equipment (DTE) with appropriate control circuitry and algorithms shares the use of the medium and available transmission. With ring topology, the network cable interconnects DTEs until they form a ring. A feature of a ring topology is that there is a unidirectional operation with a direct Point-to-Point (PP) link between each neighbouring DTE. Again, appropriate medium access control algorithms ensure that the use of the ring is shared between the community of DTE. A variation of the bus and ring, known as the hub topology, has the appearance of a star topology; however, the hub is simply the bus or ring wiring collapsed into a central unit. The wires used to connect each DTE to the bus or ring are then extended out from the hub. This is unlike a Private Digital Exchange (PDX), as the hub does not perform any switching functions, but simply consists of a set of repeaters that retransmits all the signals received from the DTEs to all other DTEs, in the same way as the bus or ring network.

The various network topologies all require medium access control methods to allow multiple DTEs to communicate without data collisions. With a star network the central controlling element (a PDX, for example) ensures that the connection between the two communicating DTEs is reserved for the duration of the call. However, with both ring and bus topologies there is a single transmission path that links all the DTEs together. Consequently, a discipline must be imposed on all DTEs connected to the network to ensure that the transmission medium is share between all DTEs. The two techniques that have been adopted for use in the various standards documents are Carrier-Sense-Multiple-Access with Collision Detection (CSMA/CD), for use with bus network topologies, and control token, for use with either bus or ring networks. An access method based on a slotted ring is also widely used with ring networks.

The CSMA/CD is using solely with bus networks, with this topology all DTEs are connected to the same transmission medium and thus the medium is said to operate in a Multiple Access (MA) mode. All data transmitted is framed up with the destination DTE address at the head of the frame. The frame of data is then transmitted or broadcast across the transmission medium. All DTEs connected to the transmission medium detect that a frame of data is being transmitted. When the require destination DTE detects that the current frame has its own address at the head of the frame, it continues to read in the data and then responds according to the defined link protocol. With this style of operation it is possible for several DTEs to transmit a data frame over the same transmission medium at the same time, causing data from all sources to be corrupted. To reduce the possibility of this occurring the DTEs listen - electronically - to the transmission medium, then if a carrier signal is sensed (CS), the DTE defers its transmission until the passing frame has been transmitted. There is still a possibility that two DTEs wishing to transmit a frame may simultaneously determine that there is no activity on the bus, and hence both start to transmit their frames of data simultaneously. This simultaneous transmission causes a data collision, resulting in the contents of both frames being corrupted.

The electromagnetic propagation speed along a cable is 200,000 times faster than the propagation of sound in water, making the data frame significantly longer in time than the time to propagate between DTEs. This high propagation speed allows a CSMA/CD system to function with minimal data collisions. However, to operate such an access method in a medium that has a relatively low propagation speed is not feasible because the propagation time is comparable to the data packet time.

Another way of controlling access to a shared transmission medium is by the use of a control token. The token is passed from one DTE to another according to the defined set of rules understood and adhered to by all DTEs connected to a medium. The access method operates by only allowing the DTE with the token in its possession to transmit a frame of data. Once the DTE in possession of the control token has finished transmitting it passes the token on to another DTE to allow it to access the transmission medium. Monitoring functions within the active DTEs connected to the network provides the basis for initialisation and recovery for both the logical ring and the control token.

An alternative access method for controlling a ring network is the slotted ring. The ring is intialised by the monitor (a special node) to contain a fixed number of binary digits. This stream of digits continually circulates around the ring from one DTE to another. A DTE wanting to transmit waits until it receives a stream of digits with the Full/Empty flag in the empty state. The DTE can then load up a fixed frame width of data into the circulating stream and set the Full/Empty flag to indicate that the stream holds data. The stream also holds the destination and source address, along with some control bits. When the destination DTE receives the frame it reads in the data and continues to circulate the unmodified frame. After reading the frame contents, it modifies the pair of response bits at the tail of the slot to indicate that it has read the contents of the frame, or alternatively, if the addressed DTE is busy or inoperable, the response bits are marked accordingly. The DTE that transmitted the original frame waits until the frame has circulated the ring by counting the number of slots that are repeated at the ring interface. Upon receipt of the bit of the slot used to transmit the frame, it marks the slot as empty and waits to read the response bits from the tail of the frame to determine what action to take next. The monitor-passed bit is used by the monitor to detect whether a DTE has failed to free up a slot after it has transmitted a frame. The disadvantages of a slotted ring access control are the vulnerability of the monitor node as it is required to maintain the basic ring structure and the limited data size of the slots. Hence transmission often requires multiple slots, whereas with a token ring, a DTE in possession of the control token can transmit a complete frame containing multiple bytes of data as a single unit.

The various network topologies discussed above are generally used to connect computers together in controlled and static conditions, which is the case for seabed transponders. However, the mobile transponders are a different problem as they can be introduced after the logical ring has been established, hence preventing the unit from communicating to the other units and vice versa. Also, should a unit be removed the logical ring will be broken and the control token will be lost. Although the active DTEs connected to the network monitor the token and provide functions to recover the token and re-initialise the logical ring. Computer interconnection systems that have similar inherent problems are InfraRed (IR) or Radio communication, i.e. wireless LANs. A comparison of IR, Radio and Acoustic is shown in Table 4.1

The communication link between stations is purely acoustic as the units are randomly separated with no electrical connection. This situation is very similar to a laptop computer communicating with a mobile telephone or a printer using an Infra Red (IR) light beam. The laptop computer has no physical connection to the mobile telephone or the printer, so the computer has to discover what is within communication range by initiating a discovery command. To ensure that various devices can communicate with each other there has to be a communication protocol. The IR communication protocols have been generated by the Infrared Data Association IrDA, which is a collaboration of the main international IR communication manufacturers, namely: IBM Corporation, Hewlett-Packard Company, Apple Computer, Inc., Counterpoint Systems Foundry, Inc. There are several layers to the IR communication protocols, but the one that is of interest is the Serial Infrared Link Access Protocol (IrLAP). IrLAP is the specification intended to facilitate the interconnection of computers and peripherals using a direct half duplex serial infrared physical communications medium such as that provided by the IrDA serial infrared physical layer. It describes the functions, features, protocols and services for the interconnection between computers at the data link layer. The data link layer protocol is based on pre-existing standard asynchronous High-level Data Link Control (HDLC) and Synchronous Data Link Control (SDLC) half-duplex protocols as used on multi-drop links. For a three-dimensional positioning system to function there must be a minimum of four stations, the fourth unit being monitored or positioned by the seabed array. The communication protocol needs to allow several stations to communicate without inter-station interference, hence Point-to-Point (PP) and Point-to-Multi-point (MP).

Property	**Acoustic**	**InfraRed**	**Radio**
Multipath Fading	Yes	No	Yes
Multipath Dispersion	Yes	Yes	Yes
Source of Bandwidth Limitation	Choice of Projector & Physics of the medium	High photodiode capacitance, multipath dispersion	Regulatory
Source of Dominant Noise	Ambient background, Man-made	Ambient background Light	Interference from other users
Security	Low	High	Low
Range	Medium	Low	High

Table 4.1 Comparison of the various communication channels

The IrLAP protocol is complex and the data transfer is too intensive to integrate into the underwater stations completely, hence the development of the Acoustic Link Access Protocol (ALAP), which has been based on IrLAP and the various other network topologies discussed earlier. One of the main problems with underwater communications is the propagation speed of sound in water or the 'transit-time' of a data frame from the transmitting station to the receiving station and the requirement in this situation for omni-directional transmission. Whereas, in IR communications the transit-time can be assumed to be negligible due to the propagation speed of light and the operating range of IR equipment. For example; consider two communication cases, one being in-air using IR and the other in-water using acoustic; in both cases the two stations are separated by ten metres. The transit time of a data frame using IR is approximately

33ns, whereas in the acoustic case the transit time is about 6ms. The ALAP introduces several parameters to ensure the time of flight is accounted for.

IR links can be design to operate in two different modes; directed and non-directed. The non-directed infrared links do not require alignment between the transmitter and receiver and can be categorised as either line-of-sight (LOS) or diffuse; a LOS link requires an unobstructed LOS path for reliable communications. Whereas a diffused link relies instead upon the diffuse reflections from the ceiling or other reflectors, typical office materials such as painted surfaces, wood, carpets reflected typically 80% of the infrared light. As with radio and acoustics communications, IR experiences similar multipath propagation, causing the amplitude at the receiver to undergo severe amplitude fades on the scale of a wavelength, so that a detector that is smaller than a wavelength experiences multipath fading.[4.2] However, the relative size of an infrared detector is immense, typically 10,000 square wavelengths. The total photocurrent produced by a photodetector is proportional to the integral of the squared-electric field over the entire photodetector surface. The large surface area thus provides an inherent spatial diversity, preventing multipath fading. In acoustic terms, the photodetector can be viewed as a large two-dimensional array of miniature square-law elements with direct combining. Although there is no multipath fading, the multipath optical propagation does produce dispersion.

The main interest areas of the IrLAP protocols are:

- Extended addressing to account for the mobile, *ad-hoc* nature of the medium.
- Dynamic address conflict resolution procedure.
- Dynamic station discovery and identification procedures.
- Connection set-up to include a negotiation framework, which stations use to establish the best connection characteristics that both connecting parties can support, i.e. maximum packet size, frequency and encoding technique.
- Any station can contend to become the Master station.
- Medium access rules to resolve contention between stations competing for control of the medium and to prevent hidden node transmissions.

The above IrLAP services have been designed into the ALAPs, however overall most of the IrLAP service have been removed or reduced in complexity to minimise the data transfer between stations, as the data rate of the acoustic communication link is typically slower than most IR links.

4.1.1 Unbalanced Data Link

As with the IrLAP the ALAP treats the acoustic medium as an unbalanced data link due to its half duplex nature, lack of collision detection, variable communication frequency/speed, and the unpridictable nature of the environment.

An unbalanced data link involves two or more participating data stations. For control purposes one station on the data link assumes the responsibility for the organisation of data flow and for unrecoverable data link error conditions. The data station assuming this responsibility is known as the Master or Master station, and the frames it transmits are known as command frames. The other stations on the data link are known as Slave or Slave stations, and the frames they transmit are known as response frames.

All transmissions over an unbalanced data link go to or from the Master station[7]. A communication link can be established between two other stations upon the explicit permission of the Master, and only to perform a task requested by the Master. However, a Master station can relinquish its control of the medium to one of the other stations. For the Master to relinquish control to a Slave transponder after positioning of the mobile units has commenced requires the Master to communicate vital co-ordinate data and transponder IDs to the new Master unit. Failure to transmit all of the seabed transponder positions will result in a datum shift, hence positions calculated using the new datum would not be consistent with the old datum. For the above reason, once positioning of the mobile units has commenced the Master only relinquishes control if it receives a command from a surface or diver unit, hence only upon user intervention. There is only one Master station; all other stations must be Slave stations. Not all stations must have Master capabilities, but those that do not, can only communicate with stations with

[7] There is a timing response command in which the Master station instructes a Slave station to communicate with another Slave station and relay the response back to the Master station.

Master capability. In this design, all the transponders have Master capability; however in certain applications it may not be necessary.

4.1.2 Modes

ALAP data stations can be in either of two modes: Normal Response Mode (NRM) or Normal Disconnect Mode (NDM). These correspond to the connection state and the contention state respectively. Each station, after entering NRM, knows which role it is to play: Master station or Slave station. When in NRM, stations are operational and connected, and when in NDM they are operational and disconnected.

When in NRM, a Slave station will initiate transmission only as a result of receiving explicit permission to do so from the Master station. After receiving permission, the Slave will initiate a response transmission. Communications in NDM are contention-based. As a result, stations are not allowed to transmit except when initiating a discovery command and hence in a Master state. This differs slightly from IrLAP as they allow contention-based communication, if the unit abides by the NDM media access rules. However, due to the propagation delays associated with acoustic communications these rules could not be sensibly ported to the acoustic protocols.

Although contention based communications are not allowed, the baseline acoustic marker beacons only transmit, and hence the beacon transmissions are unsolicited. Ideally, the beacon transmission will be in a different frequency band, enabling the transponder to disable or enable beacon reception.

4.1.3 Bit and Byte Ordering

Each byte is composed of 8 bits, numbered, 0 to 7, where 0 is always the least significant bit (LSB) and 7 is the most significant bit (MSB). Bytes are represented throughout in the following forms:

- Binary form - a byte is represented as a sequence of 8 digits (1 or 0), with the least significant bit on the right and the most significant on the left.

- Hex form - a byte is represented with two hex-decimal digits with the least significant nibble (4 bits) on the right and most significant on the left.
- Diagram form - a byte is represented as a rectangle with slots for each bit. The leftmost slot contains the most significant bit and the far right slot contains the least significant bit.
- Multiple byte form - this is represented as a rectangle with slots for each byte. The least significant is on the left and the most significant is on the right.

Examples of each representation for the hex value X'F0' are shown below. The multiple byte shows three bytes sequence of X'F0', X'F1' and X'F2':

Binary form - B'11110000'

Hex form - X'F0'

Diagram form -

7							0
1	1	1	1	0	0	0	0

Multi byte form -

1 byte	1 byte	1 byte
X'F0'	X'F1'	X'F2'

4.2 Data Packets & Framing

All of the network topologies discussed transmit data in packets and requires framing data such as, destination/source address, status flags, error flags etc. The ALAP is based on the various network protocols discussed and the similarities will become obvious throughout the rest of this chapter. The example frame of data shown in Figure 4.2 Frame of data, is a 16-bit data transfer between unit *from_id* to unit *to_id,* the command *cd* indicates to the receiving unit the type of command packet. As the command packets are not a fixed size the command bits enable the receiving unit to extract the data correctly. The above data frame does not show the packet framing that is used for multiple packet transmissions or the synchronisation cycles and start-bit. The synchronisation and start-bit are discussed in Chapter 5.

cd	to_id	from_id	data_int 0	crc

Figure 4.2 Frame of data

The 8-bit Cyclic Redundancy Check (CRC or *crc*) appended to the end of the data packet enables the receiving transponder unit to check if it has received an error-free packet. The data packet error control is also discussed in Chapter 5. The three-bit command *cd* gives eight Master command/response data packets used for the majority of communications. The eight Master commands are shown in Figure 4.3 , the commands vary in length (number of bits), but the first byte of all packets holds the command *cd* and the destination address *to_id.* All of the commands have the source address *from_id* following the *to_id,* with the exception of the discovery command, which does not need to include this information because only the Master is allowed to transmit this command and the Master's ID address is predefined.

The command database is expanded using the extended command, which facilitates up to 256 commands or responses (see Appendix C, Extended Commands). These special commands enable the transponders to transfer multiple data packets, request battery status information, transmit reset commands, etc. Some of the special function extended commands have not been defined and are intended for expansion purposes and user-defined instructions.

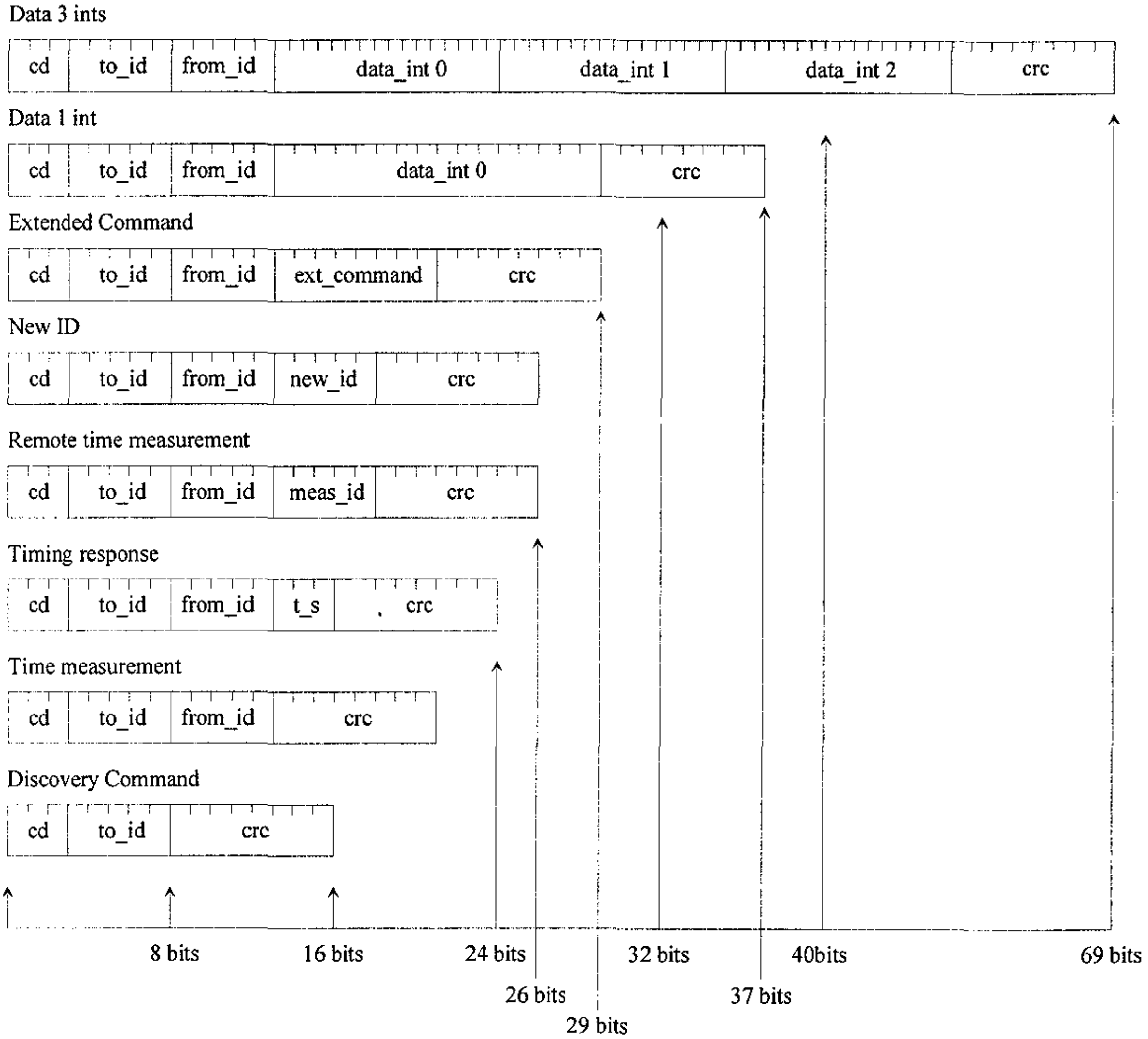


Figure 4.3 Command Database

4.3 Start-up States and Services

Considering Figure 4.1 and the general concept of the positioning system, it is apparent that there are two distinct operating periods: calibration (NDM) and positioning (NRM).

The calibration period controls the start-up conditions of each unit, up to the point where there is enough information known to enable the seabed array to begin positioning the mobile units. As the system is designed to be totally dynamic, even the seabed units are initially considered to be mobile units. With reference to Figure 4.4 , the following description gives an overview of the calibration period.

The transponders are considered to be in NDM operation until either they receive a discovery command or the random start-up timer expires and the transponder assumes Master status.

- **Start up (NDM)** - Upon power-up the processor module configures itself and initialises the peripherals to the predetermined initial values. The unit then burst-reads in a buffer of data to evaluate the noise level and dc-offset of the ADC. The noise level value measured is used to adjust the interrupt level and is also used as a random seed for the random number generator[8]. A random delay timer is then set, based on a value returned by the random number generator (between 10 - 60 seconds) and the input interrupt is enabled. Thus the unit is now listening for a discovery command.
- **Master Discovery (NRM)** - This state is entered when the random delay timer expires; the unit is now the Master unit and controls the medium. The Master transponder now starts to emit discovery commands to determine whether other transponders have been deployed. Once becoming the Master unit there are only three reasons why it may relinquish control:
    1. Two units have tried to become the Master of the medium (Two Masters conflicting).

[8] The random number generator is an eight-bit pseudo-random number generator (1 - 256 as 0 is not a valid number, due to how the random number generator functions). Random numbers referred to in this section have been generated by this random number generator.

2. The Master is a mobile unit. The Master will discover that it is the mobile unit, if two or more units are moving and the unit cannot detect any stationary units (unlikely to occur as the seabed array in most cases would be deployed before any mobile units are in the water).

3. Human intervention; i.e. a reset command is sent from a surface unit or diver unit.

- **Slave Discovery (NRM)** - If during the random delay time a discovery command is received, the unit becomes a Slave device and enters the Slave state machine. On receiving the discovery command, the unit randomly selects a response time slot between 0 and 15 and responds as soon as the time-slot period is valid.

- **Master ID Allocation** - If during the discovery period the Master processes the Slave responses and allocates unique IDs, the response time-slot serves as a temporary ID address. If two Slave units respond during the same time-slot period the Master will either not allocate an ID or it will transmit a contention command to the temporary ID address, which will cause the Slave units to re-initialise their random seed.

- **Master Assign IDs** - The Master unit transmits the allocated ID to the Slave units that responded using the new ID command and awaits an ID confirmation command.

- **New ID Confirm** - The Slave unit confirms that it has received its new ID correctly by responding with a timing response. This not only confirms the ID as all communication data packets have to and from IDs, but also enable the Master to measure the round-trip time, See Figure 4.3 .

- **Master ID Confirmation** - Once a confirmation command has been received, the Slave unit is added to the registry of assigned transponders and a 'comm-link timer' is set-up to ensure that communication link between Master and Slave is kept 'alive'. A Slave device will reset itself if it is not 'spoken' to for a predetermined period. The unit will then have to be re-discovered by the Master and assigned a new ID. If an ID confirmation is not received, the Master will attempt to communicate with the Slave unit a maximum of five times before deleting it from the discovered list.

- During the initial discovery and assignment of IDs the Master has also been measuring the round-trip communication time. This enables the Master to establish the distance between the Slave units and itself. If more than two stationary Slave units have been discovered the Master unit will proceed to the time Measurement State to establish the precise position of all of the stationary units.

- **Master Time Measurement** - The Master performs timing requests with all of the Slave units in its registry to determine or confirm the distance measurements obtain earlier. These precise measurement confirm earlier measurements, enabling the Master to define whether the transponder is stationary or mobile (a stationary transponder cannot move more than 20 - 50mm (depending on the environment), otherwise it will be tagged as a mobile unit.
- **Master Remote timing request** - The previous two states enable the Master to measure the distance between itself and the other assigned units. However, the distance between Slave units is not known and for position fixing to be possible this distance is essential. So the Master sends a request to a Slave unit asking it to measure the distance between itself and another Slave unit and relay the time-of-flight information back to the Master unit.

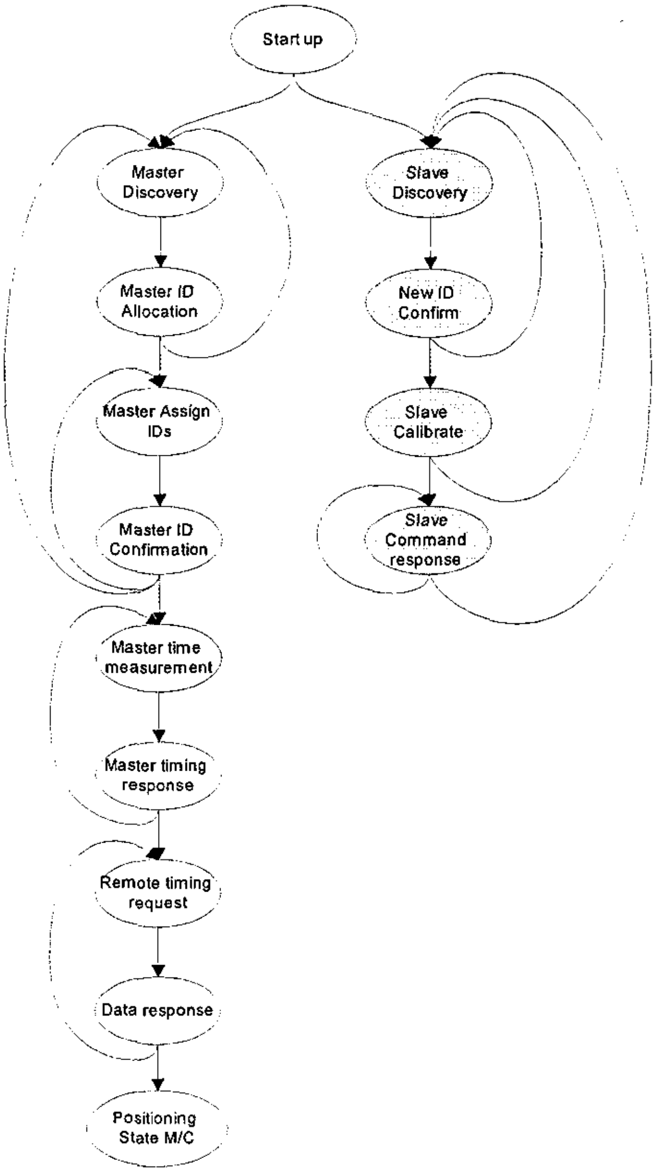


Figure 4.4 Calibration period state diagram

4.3.1 Connectionless Services

Discovery Services

The discovery procedure is used to determine the device address and some other key information of all stations that are within communication range. The station performing the discovery is called the *initiator,* and the stations that reply are called *responding* stations, or *responders.* The initiator broadcasts a discovery exchange station identification (XID) command frame, indicating a discovery procedure using 16 time slots. This frame also serves as notice of the beginning of the time slot. The beginning of the time slot is important as the *initiator* is combining the discovery and calibration tasks. All nodes that receive the discovery XID command become responders and each generates a random number between 0 and 15. If the random number generated is 0 the responder transmits a discovery Timing Response (TR) frame immediately after the constant response dead-time. Otherwise, it waits until it has timed out the random number of slot delays, and at that time it transmits its TR. The response time slot number assigns the device a temporary 5-bit identification address, this identification address is used to assign the correct device its connection address. The most significant bit of the connection address (ID) is used to indicate that the unit is assigned and prevents the temporary time-slot response addresses conflicting with active units.

The ALAP has been designed to handle up to thirty stations, hence the dynamically assigned connection address is a 5-bit number. The address field has two reserved addresses that are assigned to specific operations. B'11111' is the global or broadcast address, which is used to communicate with all stations in range and is used during the discovery procedure. B'10000' is Master address and is assigned upon becoming the Master unit.

With reference to Figure 4.5 Node's A, B, C and D are all in NDM, Node A's random timer expires first and initiates the discovery command to find all nodes that it can communicate with.

<table>
<tr><th>Time Slot
Nodes</th><th>t = 0</th><th>0</th><th>1</th><th>2</th><th>3</th><th>4</th><th>5</th><th>n</th></tr>
<tr><td>Node A</td><td>|–| XID</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Node B</td><td></td><td></td><td>|–| TR</td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>Node C</td><td></td><td></td><td></td><td></td><td></td><td>|–| TR</td><td></td><td></td></tr>
<tr><td>Node D</td><td></td><td>|–| TR</td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
</table>

where XID = Discovery Command & TR = Timing response

Figure 4.5 Response time slots

The initiating XID command indicates the start of the discovery period. The responders randomly select a time slot and respond after such period has expired. The initiator continues to transmit XID command until the minimum number of stations required for position fixing to be instrumented are connected. Once the minimum number of units has been discovered the discovery procedure is performed less frequently, as the units are now performing positioning tasks.

Address Conflict Resolution Procedure

The address conflict resolution procedure is used when two or more stations that are in range are determined to have selected identical response time-slots. The address conflict resolution procedure is used to inform the stations of the detected conflict and to guide them in the selection of new addresses that do not conflict. The response time slot acts as a temporary identification address, which the Master station uses to inform the conflicting stations. The only difference with the address conflict resolution procedure and the discovery procedure is that the discovery command XID frame is not broadcast, but is sent to the conflicting device address. This in effect multi-casts to all conflicting stations. A Slave unit, upon receiving a non-global discovery XID command, selects a new address/time-slot and responds within the discovery response frame.

Two Master Conflict

Although on start-up all units generate a random seed from the measured noise level, there is still a possibility that two units could try to become the Master unit at the same time. If a Master unit receives a discovery command during its own discovery period the unit performs a reset, so in effect relinquishes its control. If the Master has been functioning as the Master for some time and has assigned IDs to Slave units the Master will attempt to send a Master reset command to the other Master. This reset command has the duration that the unit has been operating as the Master encoded, which enables the longer serving Master to remain in control. However, if the two Masters never detect the other Master's transmission, the system would have serious problems and this can occur should the Masters be synchronised, hence transmitting a discovery command at exactly the same time, or within the allowed dead-time period after a transmission. To obviate this problem the inter-discovery time period is randomised slightly and the random seed is occasionally updated with a new noise measurement.

4.3.2 Connection-Oriented Services

Once the discovery procedure has been implemented and the various stations have been allocated connection addresses, the stations are said to be in the NRM mode. In this mode there are procedures for information exchange, reset, relinquish Master role, address re-allocation, configuration and disconnection. Each station can be addressed individually by the Master station using the unique connection address issued during the discovery procedure. The Slave devices all have time-out timers that expire if the individual unit is not addressed within a specified time period. This is referred to as keeping the transponders 'alive'. The Master transponder schedules a communication packet to each Slave device every 60 seconds. This command can be anything as long as it is individually addressing the transponder, hence global commands do not reset the 'alive' timers. Upon a Slave unit receiving a command targeted at it, the Slave unit resets its 'alive' timer (*comm_timer* - nominally 200 seconds). This ensures that a Slave unit will allow itself to reset and be re-discovered should the Master unit 'lose' the Slave unit from its registry. If this timer was not incorporated the Master unit could end up with 'dead' transponders, that is, Slave transponders that think they are 'alive' but only responds to communications with their address in the header. Therefore, should a Master have removed the transponder from its registry it would no longer be able to use that unit for the duration of deployment.

Information Exchange

Information can be exchanged to and from the Master station by any of the other stations. The Master station will initiate the data transfer by addressing the individual station using the connection address. The two basic data commands transfer either 16 bits or 48 bits of data. The data responses are often used when the Master requests remote timing information from a Slave unit. The Slave communicates the distance information in high precision calibration mode as raw time data in hundreds of nanoseconds (100ns) and transmits a DWORD (32 bits unsigned), hence a maximum time of 430 seconds. In other modes, distance is conveyed in 16-bit resolution (centimetric resolution), enabling x, y & z information to be transferred in a single packet.

The 48-bit data packet is used to communicate the co-ordinates of the seabed transponder to the mobile transponders following the calibration procedure. The *data_3_ints* command is sent from the Master, but the Master adjusts the *from_id* address to indicate the transponder to which the co-ordinates are assigned. This is only allowed following an extended command issued by the Master (co-ordinate transfer command) that configures the Slave units to expect transponder co-ordinate data. This state is optional but it is ideal, since all the transponders know their own co-ordinates and those of all the other seabed units relative to the Master. After each co-ordinate transmission the Master waits for an extended command/response from any transponder that did not receive the co-ordinates without error or within the required time period following the extended co-ordinate transfer command. Should a response be received the Master re-transmits the data; after three attempts the Master tags the unit(s) to indicate that they have not received the co-ordinate data. Also, new stationary units that join the seabed array in the post-calibration period are allocated a unique ID as usual and once the position of the new transponder is known, its co-ordinates are transmitted globally following an extended co-ordinate transfer command.

Reset (Global/Individual)

The reset function is so that under extreme circumstances the seabed transponders can perform a complete 'reboot' of the system. The command can be issued to an individual transponder or globally to all seabed and mobile transponder in range. When this command is issued, each transponder enters the NDM, then as if they had been re-deployed the discovery procedure is initiated by the transponder that times out of its pseudo random time delay first.

Relinquish Master Role

A command that enables the Master station to relinquish its Master role can be issued by the Master station itself or another station that has a user interface, i.e. the surface vessel or a diver unit. The Master station can relinquish it role for several reasons; to allow a preferred positioned transponder to take control, to allow other transponders to communicate with each other rather than via itself, hence reducing the data transfer rate. There is also the option to allow the diver or surface vessel to select the more appropriate control transponder, e.g. before deploying more transponders to increase the survey area.

Although this command is available there is a significant overhead when used if the same position datum is required. For this reason it can only be initiated by a user-interface transponder, with the command stating whether the old positioning datum is required when re-initialised. Hence, if the old datum is not required, transponder co-ordinates and IDs do not have to be relayed.

Address Re-allocation

Address re-allocation by the Master transponder may be necessary to clear a conflict or to allow for additional transponders. The command to change an assigned transponder's ID address is exactly the same as when issuing IDs in the first place, i.e. *new_id* command. The command is sent to the transponder (*to_id*), with the new ID (*new_id*) as during the discovery period, the transponder then responds to confirm its new ID. The response has to be received by the Master; if it fails to receive a response the Master will assume it has not received the new ID and will attempt to re-assign it again. The Master will re-try up to five times before removing it from its registry.

Communications Optimisation

This is the process of configuring the Slave transponders by the Master transponder to alter the communication link between them. This option enables the Master transponder to change communication frequency, inter-packet period, modulation and power level etc. thus increasing the data transfer rate and reliability of the communication link. These changes, before being implemented, must be confirmed by the Slave unit before the Master will attempt to communicate using the new settings. Each transponder is a structure (C code *struct* is a collection of data, possibly of different types, grouped together under a single name for convenient handling), and within the structure is information about each of the transponders, which allows each transponder to be communicated to in a different and more efficient format.

Disconnect

The disconnect command is very similar to the reset command; however, the unit will not attempt to become the Master unit if a disconnect command has been issued. The command causes a Slave transponder to enter the NDM, at which point it can re-connect to the system when a discovery XID command has been received from the Master transponder. A Slave transponder can disconnect itself from the Master, if for example, data for the unit is not being received, which could be a problem with the Master station 'losing' the Slave from its registry or transmitted data could be being corrupted.

Position fixing

Normal position fixing mode is when the Master addresses a mobile unit instructing it to transmit a global timing response command. The seabed units choose a time response time slot (t_s) and respond, the difference being that the *to_id* is the mobile unit originally addressed by the Master. This procedure is relatively slow, as there are three discreet communications to enable the mobile transponder's position to be calculated. Step 1: Master instructs the mobile unit to transmit a timing response command. Step 2: Mobile unit transmits timing response command. Step 3: Seabed units respond, giving the mobile unit 'x' slant ranges to itself. Also, the communications from the Slave devices allows the Master to position the mobile unit. If the mobile unit does not have co-ordinate information of the Slave units (which is transferred globally during the calibration period) the Master transmits the (x,y,z) position data to the mobile unit and more importantly to the surface vessel and other diver units. The nominal update rate, which is dependent on the dead-time and turn-around time-slot periods, is about 3 to 4 seconds. If there are four divers in the water and assuming a sequential update of their positions, the time between position fixes would be greater than 15 seconds. This period does not include the Master's task such as periodic discovery and 'alive' counter reset communications.

Fast Position Fixing (1)

In this state the Master transponder maintains control, but a Slave unit can be allowed a Fast Track option. In this case the Master station allows all units to respond to a particular transponder without communicating via the Master. This increases the update rate of a unit's position, which may be required during precise measurement exercises, or for a fast moving AUV or ROV. This fast position fixing state is used when there are few mobile units connected and the Master can allow the mobile unit to function as a temporary Master unit. The fast-track command is sent out to all transponders, and the transponder identified in this command can now issue timing response commands. The Slave seabed units that received the fast track command will now generate timing responses to the fast track ID transponder for the number of times the Master has allowed. During fast positioning each seabed station responds in a pseudo-random time slot (t_s, 3-bits), of which there are eight. Due to the duration of these slots it is possible for the mobile unit to receive two responses in the same time-slot without data corruption, however there is a possibility that timing response command from the seabed unit will be corrupted due to a neighbouring transponder responding in the same time slot. The possibility of corruption also depends on the position of the mobile unit, relative to the seabed transponders Assuming four seabed transponders the likelihood of response collisions is relatively high, but the increased positional update rate negates the unrecoverable collision error occurrence. If the collision error is too great to give the mobile unit the required position update rate then an extended timing response can be used. This allows up to 32 timing response slots, hence reducing the probability of packet collisions from a 1 in 2 chance to a 1 in 8, for the previous example. There is a maximum number of fast-positioning responses (time period) allowed before the Master resumes control. The fast-track transponder can finish a fast-track mode at anytime by issuing a fast track termination command, which causes all of the transponders to abort the fast-tracking state. Also, any Master command will cause the Slave transponders to abort the fast-tracking state.

Fast Position Fixing (2)

In the fast position fixing 2 state, the Master transmits a synchronisation command, which all of the stationary transponders respond to in random time-slots. This configuration can be used when there are a reasonable number of mobile units requiring positioning update. The overhead of this state is that the co-ordinates of all of the seabed units have to be communicated to the mobile units prior to the state being entered and also there has to be redundant transponders, i.e. at least four seabed units. This is because the Master acts as a synchroniser and the mobile units (Slaves) receive the synchonisation command followed by x number of Slave seabed transponder timing responses. Chapter 7 explains the principles behind the various positioning options are implemented.

Emergency Situation

The regular discovery period during normal operation to detect new transponders, also presents a window for active units (units assigned IDs, hence 17 - 31) to transmit to the Master urgent information. This window is generally for emergency transmissions, in which a mobile or seabed transponder can relay urgent information to the Master. This could be a situation in which a mobile diver unit has got into difficulty and the Master unit has not received the distress signal of the mobile unit. Vital information is conveyed to the Master to stop general positioning and focus on the distressed diver. The Master upon receiving a distress signal transmits an emergency extended command that configures the Slave unit into the correct state, for example if the diver unit has a graphical interface then the stricken diver's position would be indicated. The Master transmits globally the (x,y,z) co-ordinates of the distressed diver to enable the surface unit and diver units to locate the stricken diver quickly. This emergency state is show in Figure 4.6 main state machine overview. The emergency situation was invoked in the test tank to observe the transmission of the emergency command and the global transmissions of the stricken diver's position (a mobile unit).

4.4 Operating Procedure

1. Deployment of Transponders

The transponders should be deployed at sensible locations to achieve the best performance from the positioning system. For example, deploying the transponder array in a line will degrade the position accuracy of system.

2. Discovery and Address Allocation

The randomly assigned Master transponder will automatically allocate individual addresses to the Slave transponders. During address allocation the acoustic response packets are still transmitted at precise times, enabling the Master to measure the distance to the Slave transponders.

3. Array Calibration (Master)

The precise distance between each transponder is then measured by an exchange of acoustic commands initiated by the Master transponder. The individually addressed Slave transponder respond in a minimum turn-around-time to reduce timing errors due to clock drift.

4. Array Calibration (Slave)

The above task measured the distance between the Master and the Slave transponders (t_{m1}, t_{m2}, t_{m3}, etc). The Master now instructs the Slave transponders to measure the distance between set pairs of Slave transponders and telemeter the propagation time back to the Master. This completes the calibration of the seabed transponder array, but during this period the transponders must have remained stationary. If the distance to a seabed transponder changes during the array calibration procedure then the unit will be tagged as a mobile unit.

5. Deployment of Baseline Marker Beacons (Optional)

The baseline marker beacons can be deployed at anytime, but the transponder array will now only enable marker beacon reception. To position the marker beacons in two-dimensions the distances between at least three transponders in the seabed array must be known. The time difference in the arrival of the unsynchronised transmission of a beacon is measured by relaying the arrival time at each transponder back to the Master. The (x,y) positions relative to the seabed array co-ordinate grid of at least two beacons are required to facilitate co-ordinate transformation on subsequent dives.

6. Mobile Positioning

The seabed array is continuously attempting to discover new mobile devices. Once mobile units are attached the discovery frequency is reduced proportionally to the number of attached mobile transponder units.

7. Task Complete – Retrieve Seabed Transponder Array

The seabed transponder array is retrieved by divers or by transmitting an acoustic release command. An acoustic release command operates a mechanical release mechanism that disconnects the positively buoyant transponder from it seabed weight, causing the unit to float to the surface. Following a release command the transponder still responds to time measurement commands, which enable the surface vessel to measure the range to it to aid its recovery. The baseline marker beacons are left on the seabed.

8. New Positioning Task – In the above survey area

Returning to a site with baseline marker beacons installed enables the data from previous surveys to be immediately correlated will new data by transposition of the calculated positions onto the fixed co-ordinate grid delineated by the seabed beacons. Steps 1 to 4 are performed as above, but step 5 does not require the deployment of new baseline markers as they are already installed. The system will position the baseline marker beacons in the same manner as above, but the positions of the beacons are relative to the new transponder array and not the previous deployment.

4.4.1 Co-ordinate Transposition

The transponders can be programmed with the previous position of the beacons with respect to the transponder seabed array, however this requires all the transponders to be programmed via the serial link at the surface. Alternatively, if there is a surface vessel, the Master can request the positions of the beacons once the current position of the beacons has been established. The Master unit can then calculate the new transposed co-ordinates of the seabed array to align the position data. Alternatively, the Master can apply the same rotation and transposition matrix to every calculated mobile position or the data can be post-processed.

4.5 Software

The software in the transponders quickly becomes difficult to manage if it is not structured well. The dynamic 'intelligent' design of the transponders makes coding difficult as the units are programmed to make decisions depending on a set of events that have just happened. The non-user interface stand-alone operation of the transponders requires the code to operate in states, depending on whether the unit is the Master or a Slave and also depending on the task in hand. The design adopted is an event driven state machine as indicated throughout this chapter.

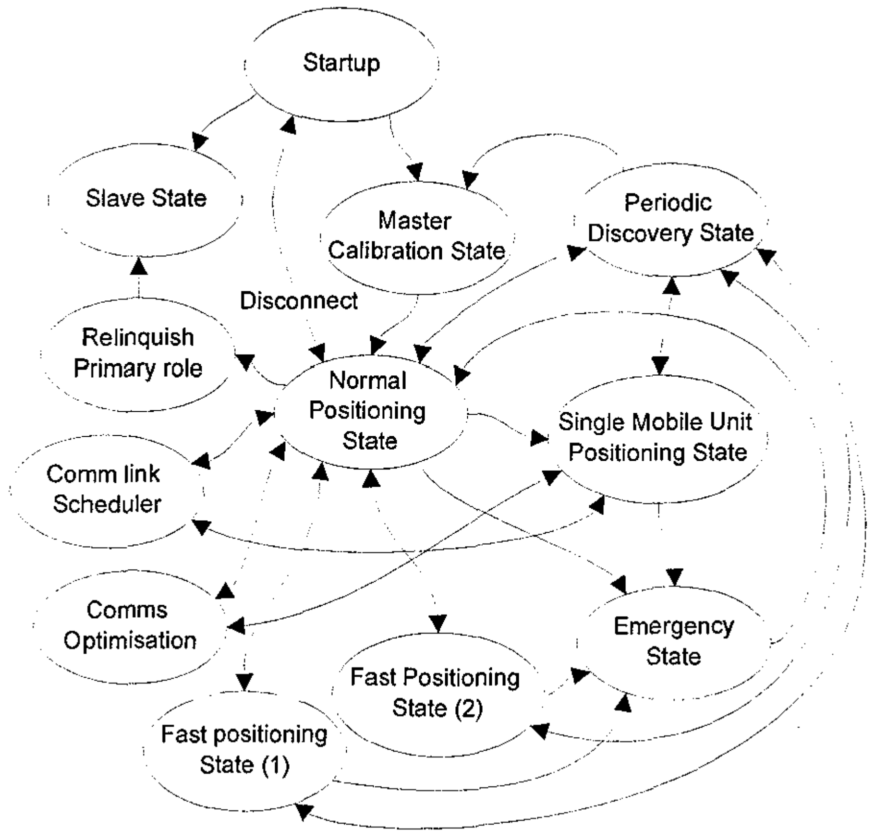


Figure 4.6 Main Software State Machine

4.5.1 Event Handler

The transponder software is designed so that when an event occurs, be it a timer expiring or a data packet captured, an event flag is set. This event flag invokes the dispatcher that calls the current state. Depending on the type of event, the state machine deals with the event (i.e. schedules a response, changes state or does nothing) and returns. The state machine can change the state in which it will be entered on a subsequent event. This state can only be changed from within the state machine itself, hence the state variable is static to the state machine module, and preventing state changes from other parts of the code. The whole system being event or interrupt driven allows the processor to 'sleep' to save power when there are no mathematical position calculations or data packet signal processing to be performed.

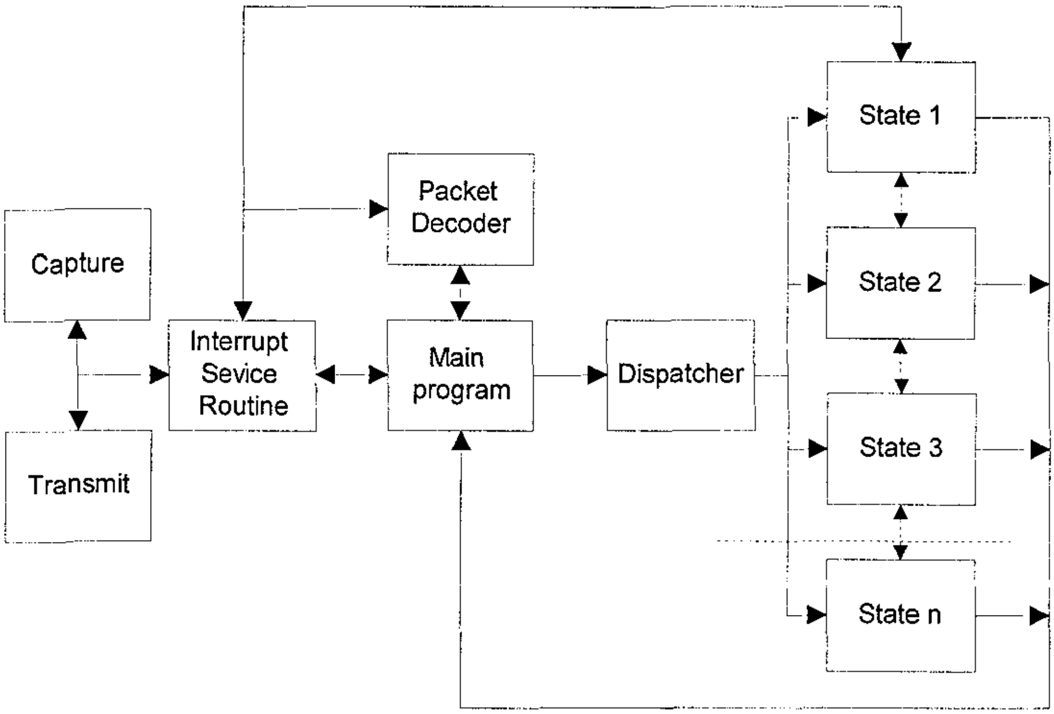


Figure 4.7 Software Basic Design

The basic software structure is shown in Figure 4.7 , in which the main program section polls (every 2ms, i.e. 1 jiffy) the captured data flag. If this flag is set then the transponder has detected the start bit of a captured packet. The packet decoder is invoked to decode the captured data packet. If a multiple packet transmission, the packet is first re-constructed into a single packet, which involves decoding and removing the framing bits.

Once the packet is re-assembled the data is decoded and then error checked (CRC) to ensure that the packet has been received error free. Once a packet has been successfully decoded a received data packet flag is set. In the main software this flag is also polled every 2ms and if it is set a routine is called to check the *to_id* address in the packet header. If the address in the packet header matches the ID of the unit or the Global address (reserved B'11111'), an event is generated. Otherwise the data packet is deleted (freed up from memory).

It is important on an embedded system to ensure that the software does not 'leak' memory, i.e. memory is allocated and then never freed up, hence the memory becomes unusable. On Personal Computers (PCs) today a small memory leak often goes unnoticed as there are 100s of Mega Bytes of RAM (or Virtual RAM). Eventually the machine will fail due to the leak, but it may take several weeks.

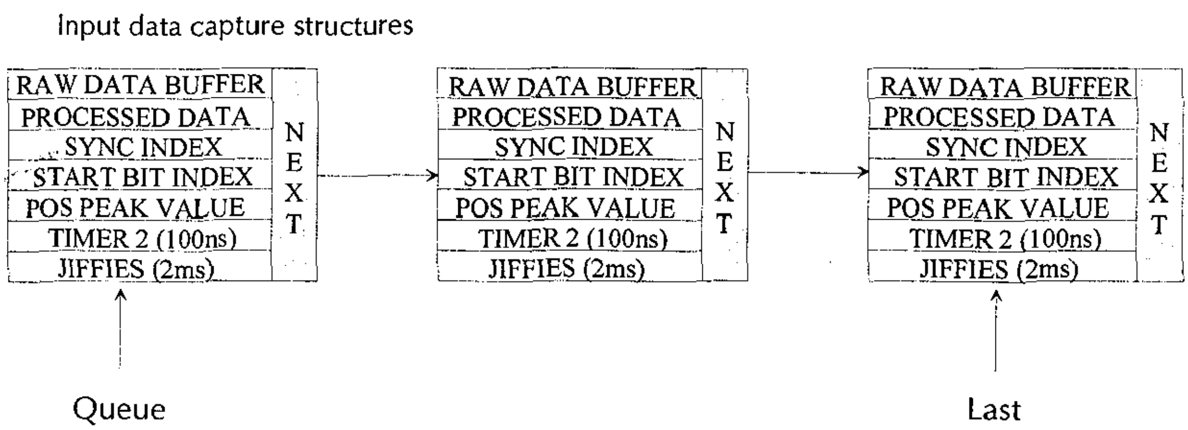
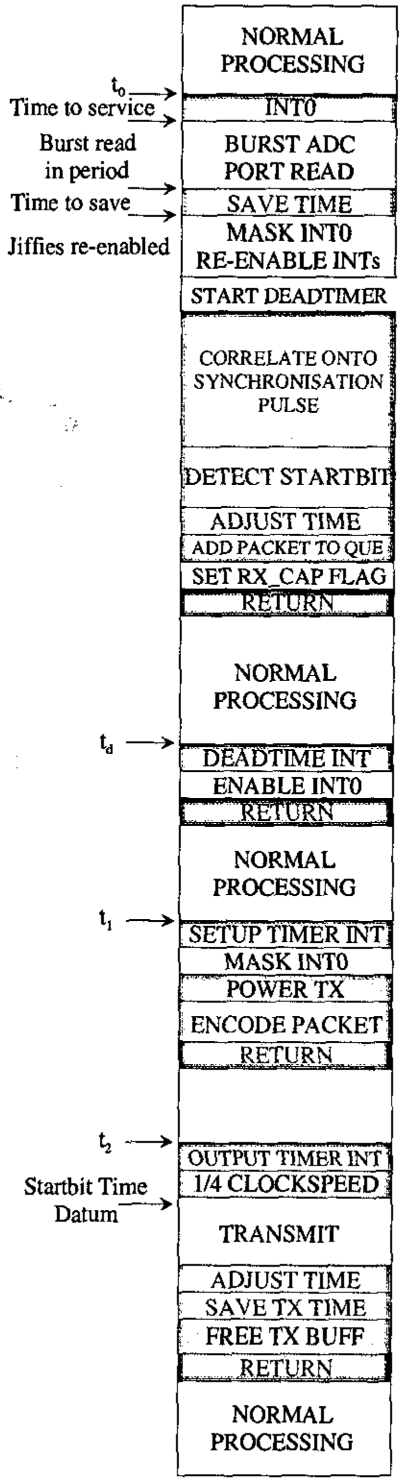


Figure 4.8 Link list of input structures

So that the system can deal with multiple packets arriving before the previous packet has been processed the system queues the packets as they are captured. The packet decoder reads the packets from the front of the queue and the capturing system appends them to the back of the queue (last). Figure 4.8 shows the link-list of input data structures. It is important that the list is managed or the list could grow out of the bounds of the available memory. Writing to memory that is out of bounds or that is not allocated is catastrophic as return address pointers, stack variables and other crucial data can be over written, which causes the system to fail quickly. Alternatively, there must be allocated memory to write to when an interrupt occurs, as there is not enough time to allocate an input structure following an interrupt, as the acoustic data or part of it will have passed during the memory allocation period. When a captured packet structure has been processed, the memory must be freed up and more importantly the queue pointer must be changed so that it points to the next structure along the link-list. That is: queue = queue->next (the Queue pointer often points to the same structure as the Last pointer).

4.5.2 Input Output Timing



The timing of the received and transmitted packets is of paramount importance, as the positioning system relies on precise timings to generate range information. All of the timings are referred to the start bit of the transmitted or received packet. The interrupt event is not the timing datum as the period to service an interrupt can vary depending on the number of variables and pointers that have to be saved following an interrupt. Also, the amplitude of the received signal can alter the point in time in which the interrupt threshold is exceeded. Systems that rely on threshold detect timing information can achieve 10 - 100μs timing accuracy, whereas timing to the start bit enables a timing accuracy that is related to the correlation accuracy. With a 5MHz-sampling rate it is possible to achieve 200ns-timing resolution. Using the start bit as a timing datum offers typically a 40-50-fold increase in timing precision. The actual time measurement precision is discussed in Chapter 5. Figure 4.9 highlights the important timing information and shows how a packet is received, processed and a response generated, all controlled by interrupt events.

Figure 4.9 Input to Output Events

At time t_0 an INT0 interrupt (envelope detector) occurs, which takes a finite time to service and then the transponder starts to burst-read in the acoustic data via the ADC.

For the period of the burst-read all interrupts are disabled, preventing timer overflows and other interrupts from being serviced.

Once the burst-read has finished the times are saved (Jiffies (2ms timer) and Timer2 (100ns timer)), hence the time at the end of the data packet is recorded. Now all the interrupts are re-enabled except for INT0, which is re-enabled upon the dead-timer expiring and is started following the re-enabling of the interrupts. The dead-timer is to prevent multiple interrupts generated by reverberation of the same acoustic data packet. The dead-time period depends up on the environment and is changed by the Master following channel impulse response measurements (if implemented).

The interrupt handler cross-correlates the start of the received signal with a waveform in RAM and tests the correlation value to ensure that the captured signal is of the correct frequency. If the correlation value does not exceed the threshold, INT0 is immediately re-enabled and the interrupt handler is exited. Otherwise, the unit then cross-correlates the received waveform with a digital representation of the startbit (can be one bit or several bits, referred to as *gold codes*, to improve startbit detection in noisy environments).

The start bit index is where the timing datum is, thus the saved times need to be adjusted to the start bit time. Knowing the sample rate, input buffer size, the start bit index and the time at the end of the buffer, the startbit time can be calculated. The interrupt routine is then exited and the processor returns to whatever it was doing prior to the interrupt (Normal Processing).

The normal processing sections are when the processor is the state machine, main program or decoding a packet, i.e. anything that is not interrupt driven. Thus, on returning from an input capture interrupt, the main program will poll the captured packet flag and call the packet decoder. The captured packet is then decoded and error checked by the packet decoder. If the packet has been successfully received then a received packet flag is set and the packet decoder is exited, otherwise the packet is deleted.

The main program polls the received packet flag and if set it then checks the *to_id* address at the head of the packet to see if it matches it own ID or the Global ID. If it matches either of the IDs, the state machine dispatcher is called to invoke the current state. The state machine then determines if the command is valid and generates the required response. The output response is not encoded within the state machine, which just schedules up data that it wants to send via the packet framer. The packet framer organises the command or response data into the correct sized packets, which correspond to the maximum packet length. The framer inserts two bits per packet that indicate to the receiving transponder what the packets are, i.e. the first packet and the last packet. The framer also resolves impossible time response issues, such as the state machine requesting that a packet was sent 30 Jiffies ago.

Each packet is then allocated two one shot timers, a set-up timer and an output timer. The set-up timer expires 30ms before the output timer. Within the set-up timer routine the INT0 interrupt is disabled and the power is switched onto the power amplifier. The 30ms allows the power amplifier to stabilise before the output transmission. The digital response or command signal is now encoded using the communication scheme that will be discussed in Chapter 5. The set-up routine is exited and the processor resumes normal processing. The output one-shot timer expires slightly before the output transmission is required. This is so the system can be placed in a known state in which the time to service an interrupt is known. Timer0 is programmed with the exact time to start transmitting, which includes the time to service the Timer0 interrupt, the exact number of instruction cycles to the beginning of the first Digital to Analogue output and number of outputs before the start bit timing reference. Only the Timer0 interrupt is enabled and the interrupt hook does not cause the processor to save the stack and pointers, hence a known interrupt latency. Once Timer0 is set up, the processor executes a halt instruction that stops the processor. Upon Timer0 expiring the output data is clocked out via the D to A converter.

The output transmission is complicated slightly by the system transmitting at a lower clocking frequency than the capture sampling frequency, which is a maximum of 5MHz. However, during transmitting this frequency is divided by 4, hence a clocking frequency of 1.25MHz. This is due to the maximum operating frequency of the D to A converter. Dividing the main processor clock by 4 also causes all of the timers to run at ¼ speed for the duration of the output. If the timers are left unadjusted the system will lose time and hence turn-around times will not be correct. Following an output transmission the timers are adjusted and the supply to the power amplifier is switched off. A dead timer is activated to re-enable INT0 after a predetermined time to prevent the unit being triggered by its own transmission.

4.6 Packet Window Allocation (PWA)

Packet window allocation is performed to enable the optimum packet length to be transferred between transponder units. The system initially captures a packet length of 10000 samples at 5Mhz sampling rate, which equates to 2ms of acoustic data when an interrupt event occurs. A basic discovery command is only 2400 samples long so the following 7600 samples provide no data as shown in Figure 4.10 . However, the system searches through the whole packet of data for a reverberation of the direct signal.

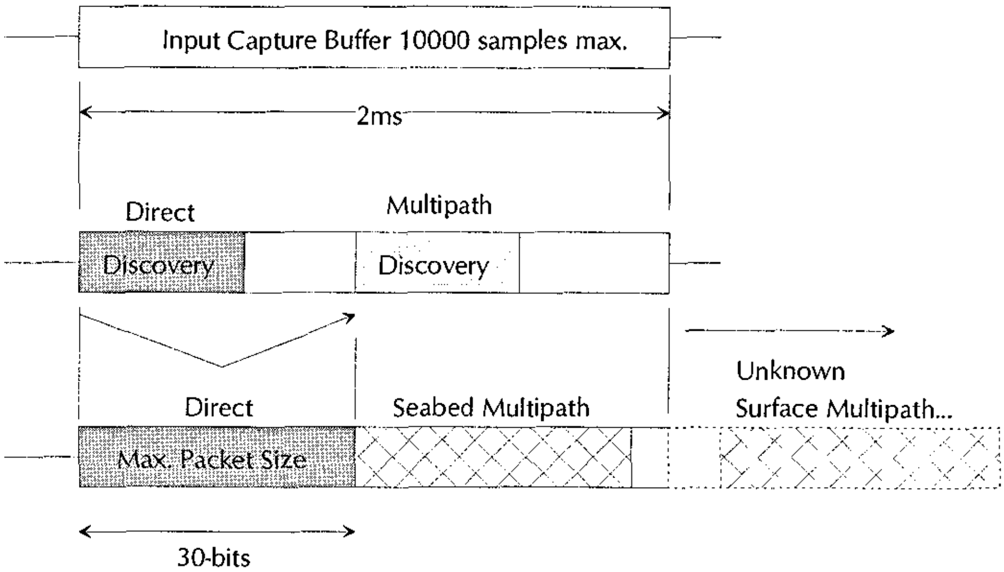


Figure 4.10 Dynamic packet length allocation

The initial search is an amplitude threshold detect, with the threshold level set at approximately 50% of the direct path captured amplitude[9]. If the signal amplitude exceeds the threshold the system attempts to synchronise and detect the start-bit as if it was the direct signal. The synchronisation correlation amplitude is used to detect whether synchronisation was achieved. If a reverberation is detected within the captured data then this limits the length of the packet. The transponder responds using the start-up maximum packet size, typically 16-bits, hence the same duration as the discovery command, 480μs. The normal timing response command is 24-bit long, so two packets have to be transferred, but the second packet will only contain the CRC. Therefore, the

[9] The signal capture amplitude is controlled by the AGC circuitry, which maintains a constant amplitude at the ADC throughout its dynamic range.

timing response to a discovery command is different to that of a normal packet, as the second packet conveys information to the master about the channel. Of the extra byte transferred, 6-Bits communicate the maximum packet length (8 - 72 bits) and the other two Bits indicate the amplitude and whether synchronisation was achieved.

The discovery timing response packet is shown in Figure 4.11 with the channel byte conveying the channel information. From the channel information the master adjusts the packet length so that multipath interference does not corrupt the data. The master then transmits an extended command *set packet length*, which is not acknowledged. Following the *set packet length* command the master sends a *new_id* command using the new packet length, which requires an acknowledgement. If the master receives the acknowledgement then the maximum packet length is stored in the allocated transponders structure. This ensures that when communicating with the various transponders the optimum packet length, which may not be the same due the layout and topography of the seabed is used. Also, this enables the slave transponder to only capture the required packet length.

cd	to_id	from_id	t_s	channel	crc

Figure 4.11 Special timing response packet

4.6.1 Multipath Tank Tests

To evaluate the systems ability to dynamically adjust the packet length, two transponders were position approximately 0.5m above the tank floor. This was to ensure that the surface and bottom multipath did not co-inside. Figure 4.12 shows a multipath measurement command, which is a packet header and start bit, and no data. The AGC circuitry attempts to maintain a full dynamic range input signal into the ADC, hence during the period when no signal is present the amplification increases as discussed in chapter 3. This is why the tank floor reverberation is the same amplitude as the direct signal.

For the test transmission shown the system precisely measures the time between the direct signal and the tank floor multipath to be 1048 samples. This was repeated several times to ensure consistent time measurements, with a maximum deviation of ±2 samples observed (20 measurement taken). This equates to a range difference of 314mm, which correlates with the approximate hydrophone positions being 2m apart and 0.5m above the seabed. In the tank the surface path also arrives at the receiver within the overall capture period, however this was ignored as the limiting reverberation is the tank floor.

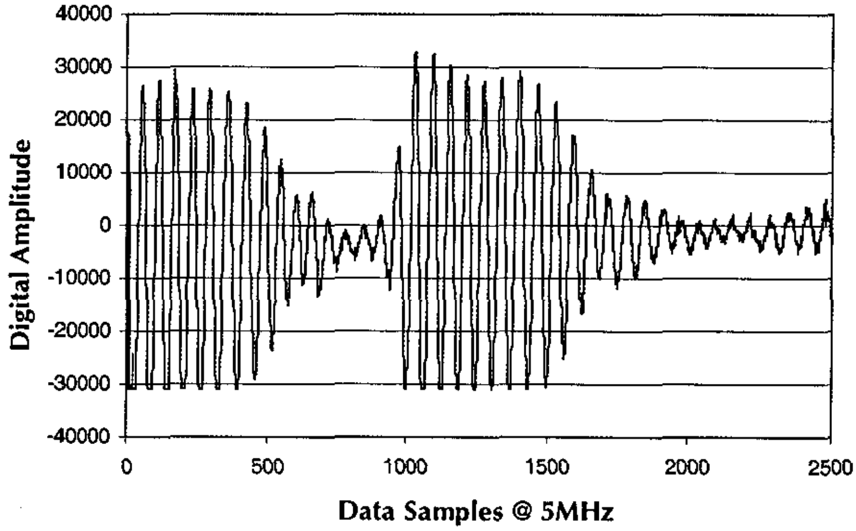


Figure 4.12 Seabed Reverberation Time Measurement Tank Tests

The AGC circuitry maximises the input signal into the ADC, but also removes the ability to assess the amplitude of the multipath signals. Relatively simple channel equalisation could be applied to enable the stationary unit to communicate longer packets. However, due to the hardware restrictions channel equalisation was not investigated further.

4.7 Protocol Testing

Designing an ALAP is relatively simple when all communication packets are received and without error - the difficult part of designing a link access protocol is assessing all the conceivable possibilities, referred to as 'what if'. The environment and low overhead communication link is bound to experience the occasional loss of a packet. It is the protocols that must ensure that the system does not fail due to a missed or corrupt packet. Initially a program was written to simulate the protocol state machine on a desktop PC and again the development period was reduced due to the upward compatibility of the embedded 186-software code. Hence, the actual state machine that was written for the embedded system could be used with minimal alterations. The Master is in control of the medium so if a command is issued to a remote Slave device, the Slave device does not have to confirm that it has received the command without error. This is because all commands require a response and confirming the correct command has been received just increases the amount of acoustic communications. The only command that the Slave unit confirms that it has received is the new ID command (*new_id*). This is for the obvious reason that when a Slave is instructed to change its address, if the command is received in error and the unit changes its address the Slave unit is no longer accessible to the Master. This new ID confirmation command multicasts as a time measurement command, as all communications are precisely timed. All commands are issued with turn around times, hence if a response from the Slave is not received within a specified period then the Master decides whether to re-issue the command or abandon it. The Master as a rule re-tries three to five times, but this is dynamic and depends on the situation. Testing the units in the acoustic test tank can be classed as an ideal test situation as all of the units can be monitored acoustically as well as debug and status messages via the serial port of each unit. Acoustic monitoring was not very useful apart from confirming that acoustic signals were being transmitted. The problem with acoustic monitoring was that it could not be determined which transponder was active and what had been transmitted. The serial port debug information was more useful as it indicated when state changes occurred, commands received, responses transmitted, timing data, positional data and registry information. So that more than one serial port, hence one transponder, could be viewed at a time an expansion card was used to enable a single computer to view three serial ports simultaneously. This, along with the structured software state machine, proved invaluable during the evaluation of the protocol design.

To simulate a higher noise environment, and hence increase the frequency in which packet errors occurred in the tank, a software routine was installed to randomly fake packet errors. The general-purpose pseudo-random number generator was used to generate a number between 1 and 256, as during start-up and responding. The pseudo random count is decremented by one each time a packet is received. Once the count hits zero a packet error is faked and the count is re-initialised with another random number. The increase in packet errors highlights states in which the system becomes unstable and either goes into a run-away or a lock-up situation. In the run-away situation the Master is continually communicating to the assigned transponders in a random rather than a systematic way. The lock-up situation is when a transponder gets into a state that it cannot get out of, although in some cases it may get out of the state by the *comm_timer* expiring. The addition of pseudo-random packet errors reduced the protocol evaluation period and helped in coding the state machine for all possible scenarios.

4.7.1 Tank test scenarios

In a real underwater positioning task there may be ten transponders in the registry of the Master transponder, five of which could be stationary seabed units and the rest mobile units. The Master transponder has to ensure that once it has discovered all the transponders it must keep them 'alive' by individually addressing them within the *comm_timer* period. This individual addressing of each transponder also ensures that the registry is up to date, i.e. a mobile unit has not left the water. The Master is also required to periodically discover if new units have entered the water and, if so, allocate an ID address and add it to its registry. The inter-discovery period cannot be too long as new mobile units that have entered the water could be without position information. This start-up period for new mobile units needs to be as short as possible to reduce the time before underwater tasks can be performed by the divers. Depending on the number of transponders already in the Master's registry the inter-discovery period is usually less than 60 seconds. This period is randomised slightly for the reasons discussed earlier. As only five units were manufactured for the research, hence allowing a maximum of four Slave units, tests included deploying each of the five transponders separately (at variable deployment times) and ensuring that the Master discovered them. Also, all the transponders were enabled at the same time to ensure that start-up time-out periods could be resolved and that the Master unit could receive four discovery responses within the sixteen discovery time slots. To increase the number of apparent transponders a test program simulated that each transponder was actually two transponders, albeit at the same position. The transponder generated two independent timing responses upon receiving a discovery command. The Master would then assign the same transponder two different IDs, to which the transponder would reply. With the simulation test program operational it was possible to test the performance of the system with eight transponders in the Master's registry. There are numerous scenarios that can occur in a real positioning system and tests such as a mobile unit leaving the water, going out of range or the batteries failing are easy to simulate in the test tank. Such tests were simulated in the tank to ensure that the Master dealt with the situation as required.

4.8 Conclusion

The protocols discussed in this chapter have functioned with up to eight simulated Slave transponders (single units acting as two independent units). The maximum addressable range of 30 transponders could not be tested acoustically due to the number of transponder units manufactured. However, simulations of the protocols and acoustic tests have shown no indication that there will be a problem. The discovery period becomes more complex as the number of units exceeds 13 (excluding reserved addresses MASTER and GLOBAL, hence 15 in total) as the normal response time slots are now also active and attached device addresses. This is resolved by the Master transmitting a reduced response time slots command prior to the discovery command that indicates exactly how many response slots are left (see Appendix B). In an operational system the unique address could be extended to 6 or even 8 bits, to allow for the maximum conceivable number of transponders. Originally, this was considered to be 16, hence the 5-bit address.

Prior to each test, the system dynamically assigns the Master unit and discovers and assigns unique IDs to all NDM Slave units, then resolves conflict issues without user interaction. The state machine, event driven design enables progressive development that does not interfere with earlier states, which simplifies the addition of new features.

The protocols discussed are totally independent to the communication scheme used at a lower level, hence enabling the system to be implemented on a frequency modulated, phase shifted or pulse position modulated communication scheme.

4.9 References

[4.1] Halsall F.
Data Communications, Computer Networks and Open Systems
Addison-Wesley Publishing Company, 3rd Edition
ISBN 0-201-56506-4, 1992

[4.2] Barry J. R.
Wireless Infrared Communications
The Kluwer International Series in Engineering and Computer Science, Kluwer Academic Publishers
pp1-181. ISBN: 0-7923-9476-3

[4.3] Zvonar Z., Brady D. and Catipovic J.
An Adaptive Linear Multiuser Receivefr for Deep Water Acoustic Local Area Networks
Proceedings of the IEEE, ISBN: 0-7803-1775-0/94
pp II-389-II392, 1994

[4.4] Green M., Rice J. A. and Merriam S.
Underwater Acoustic Modem Configured for use in a Local Area Network (LAN)
Proceedings of the IEEE, ISBN: 0-7803-5045-6/98
pp 634-638, 1998

[4.5] Spragins J.D., Hammond J.L. and Pawlikowski K.
Telecommunications - Protocols and Design
Addison-Wesley Publishing Company
ISBN 0-201-09290-5, 1991

[4.6] Xie G. G. and Gibson J. H.
A Network Layer Protocol for UANs to Address Propagation Delay Induced Performance Limitations
Proceedings of MTS/IEEE Oceans 2001 Conference, Honolulu
pp 1-8, November 2001

[4.7] Stojanovic M.
Recent Advances in High-Speed Underwater Acoustic Communications
IEEE Journal of Oceanic Engineering, Vol. 21, No. 2
April 1996, ISSN 0364-9059/96

Chapter Five

Communication Techniques

5. Communication Techniques

5.1 Introduction

This chapter discusses the problems relating to the acoustic transfer of data between transponder units. Underwater acoustic communication is difficult because of the fading multipath nature of the channel. The destructive interference, Inter-Symbol Interference (ISI), phase instability and Frequency-Specific Fading (FSF) are caused by boundary interactions of the transmitted signal. The multipath propagation characteristics of an underwater communication channel are extremely complex.

A number of different sub-sea environments can be encountered in acoustic positioning applications. The most appropriate communication technique is dependent on the channel topology; the trajectory, horizontal (seabed transponder to seabed transponder) or vertical (surface vessel to seabed transponders); the depth, shallow or deep and the operating range. The acoustic characteristics of an individual channel can fluctuate with time, temperature, weather, seabed type, surface state, salinity, sediment quantity and size, fish population and type, noise pollution and man-made obstacles,[5.1] as shown in Figure 1.2. Unfortunately, the channel for an acoustic positioning system cannot be classified as horizontal or vertical, as it will undoubtedly incur both types.

The instabilities of the channel's numerous characteristics and associated phenomena provide a medium that is not completely understood, and there is therefore scope for investigation. Even with these harsh characteristics attributed to acoustic signal propagation through the environment, underwater acoustic signalling is still the most suitable for autonomous communications. Drawing a comparison with other possible communication means, such as Electromagnetic (EM) (Optical), it can be seen that acoustic is the only viable solution. For example, EM losses in the highly conductive seawater are given by:[5.2]

$$TL = 1400 \cdot f^{\frac{1}{2}} \quad (5.1)$$

where *TL* is the transmission losses in dB/km and *f* is in kHz.

Hence, even at a relatively low frequency of 30kHz, the *TL* losses equate to 7700 dB/km, as compared to a *TL* of approximately 60dB/km using acoustics[10].

To reduce the problem of multipath interference the transmitter and or the receiver can be constructed from an array of hydrophones that focus the transmitter or receiver beam. The focusing of the beam provides high out-of-beam attenuation that reduces the intensity of surface and bottom reflections at the receiver. Also, the array can provide spatial diversity and array gain, which can increases the likelihood of detection and successful low error data communications.

The instabilities inherent in the medium combine to affect the signal characteristics predominately in the form of fluctuating signal phase and signal amplitude; thus phase and amplitude modulation schemes are particularly inhibited in the ocean environment. A number of sub-sea schemes have recently been implemented that use Phase Shift Keying (PSK) modulation techniques.[5.3]-[5.11] These schemes have required the use of multiple, widely-spaced receivers (or arrays) and or directional projectors to allow the techniques to be viable.[5.12]-[5.21] Without the ability to use widely-spaced transducers (or arrays elements) at the receiver or directional projectors, some other way of overcoming the signal fluctuations is required, if phase or amplitude-based modulation techniques are going to be used.

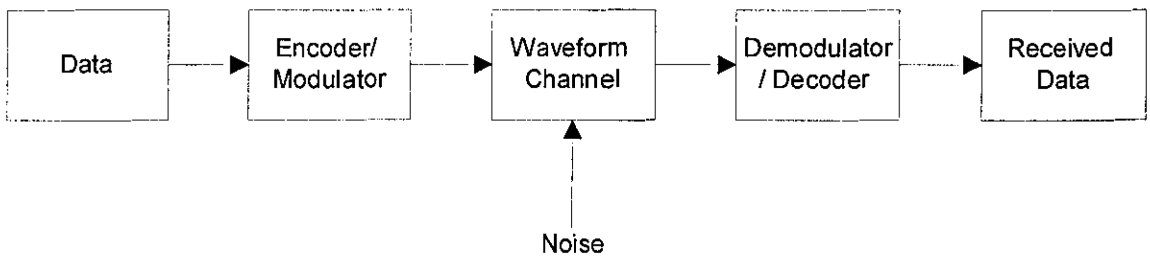


Figure 5.1 Simplified Communication Channel

[10] Assuming spherical spreading loss, i.e. 20 log R + αr, where R is the range and α is an absorption coefficient in dB/km.[5.1] pp102, 111

When designing a digital communication system there are three parameters that determine the signal energy per bit-to-noise power density ratio E_b/N_o. These are transmitted signal power, channel bandwidth and the power spectral density of the receiver. This ratio uniquely determines the bit error rate for a particular modulation scheme. In practice, the modulation scheme often arrived at can not provide acceptable data quality (i.e. low enough error performance) for a fixed E_b/N_o. A solution for changing data quality from problematic to acceptable is to use error-control coding.

There are nine basic signal-processing functions, which may be viewed as transformations and can be classified into the following groups:[5.22]

- Formatting and source coding
- Baseband signalling - Non-return-to-zero (NRZ), PCM, phase encoding, PPM.
- Bandpass signalling - coherent or non-coherent PSK, FSK, ASK, CPM, DPSK.
- Equalisation - Maximum Likelihood Sequence Estimation (MLSE), Transversal or decision feedback, preset or adaptive, symbol spaced or fractionally spaced.
- Channel Coding - M-ary signalling, antipodal, orthogonal or trellis-coded modulation.
- Multiplexing and multiple access - frequency, time, code, space and polarisation division - FDMA, TDMA, CDMA, SDMA, PDMA, respectively.
- Spreading - Direct Sequence (DS), frequency hopping, time hopping and hybrids
- Encryption - Block and data stream
- Synchronisation - Frequency, Phase, Symbol, Frame or Network.

The performance of a modulation technique in a specific communication channel can be defined in terms of the achievable Bit Error Rate (BER). Theoretical BER values can be calculated and depicted graphically as a function of signal-to-noise ratio (SNR), as is done here for White Gaussian Noise (WGN) environments.[5.23] However, a Rayleigh fading environment more closely approximates the sub-sea environment than WGN when transmitting long data packets or for continuous data transmission, even though there are significant differences between the Rayleigh case and a practical environment. Significant phenomena are neglected, including Inter-Symbol-Interference (ISI) and frequency-specific fading (FSF), therefore the theoretical results can only be viewed as guidelines.

WGN and Rayleigh fading performance are relevant to any through-water communication system as they provide an approximate model and an estimate to the performance of a particular modulation technique. The calculation of performance in a WGN environment closely approximates the situation of communication technique/protocol that transmits extremely short data packets. The design of packet transfer of data in short, high data content packets reduce the problem of multipath interference as the whole data packet is received before the arrival of any multipath signals.

Inter-Symbol-Interference is when binary symbols are transmitted through a band-limited channel, which causes the received symbols to overlap in time, giving rise to ISI. The irregularities in the channel power spectrum such as selective fading notches increase ISI beyond that of an ideal bandpass channel. If noise is not present at the output of the channel, a simple linear filter often called an equaliser can be used to reduce ISI. However, if noise is present in the system then the use of an equaliser may severely degrade system error performance.[5.24]

Rayleigh fading takes into account the signal fading effect on signal level that is generally attributed to multipath propagation. However, if the data packets are short in duration, such that fading due to multipath propagation does not occur in most situation the channel can be considered as memoryless. The proposed communication scheme typically transmits packets of data less than 1ms. So, for multipath interference to occur the direct path and the multipath must only differ by as little as 1.5m. This equates to the mobile unit being close to a boundary, i.e. the surface, the bottom or an obstacle. In situations where the multipath interference still causes problems, such as in the acoustic test tank, the transponder can reduce the packet size and transmit multiple packets. The data are then reconstructed at the receiver. Multiple packet transmission enables the transponders to function in highly reverberant environments, with packet lengths as short as 320μs (≈50cm). It is possible to reduce the packet size further, but the overhead of the synchronisation cycles and start bit becomes predominant, so the minimum packet size is artificially regulated to 8 bits of data.

Short burst packet transmission is only suitable for relatively short ranges, such as the array size discussed. When the transmission path becomes significantly longer, then this technique is not advisable, as the sound velocity profile of the water will affect the propagation time significantly for different ray paths.

The seabed transponders will often be positioned close to the seabed (within 1 to 2 metres), hence the difference in distance between the direct signal path and the bottom-bounce path is small when the transponder separation is R >> H, where R is the range between transponders and H is the transponder height above the seabed, as shown in Figure 5.2 Assuming the shortest packet duration is 320μs, for the packet to arrive at the receiver without multipath corruption, the signal path difference between the direct and the bottom bounce path has to be greater than 50cm. From Figure 5.2 it can be seen that d_{13} must be less that d_{1b}+ d_{b3} by 50cm, e.g. if the R = 14m the transponder height H has have to be at least 2m. This is an obvious restriction to the usefulness of a randomly deployable seabed array. Solutions to this problem are discussed in chapter 7.

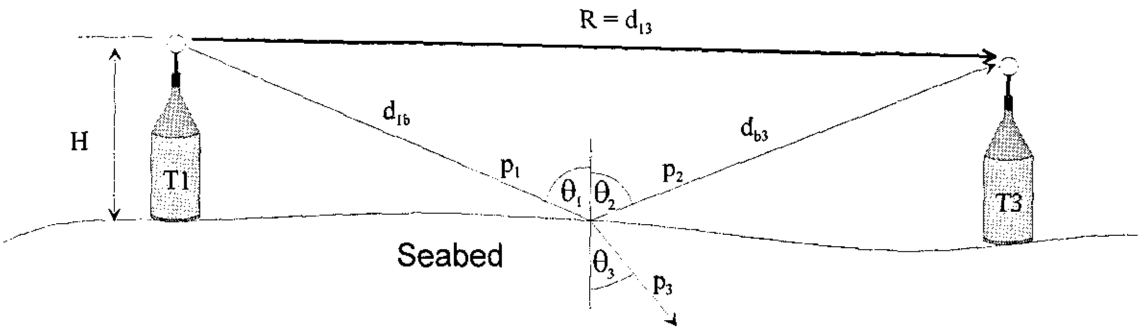


Figure 5.2 Seabed Multipath Problem

5.2 Modulation Techniques

The signal modulation techniques that can be applied to through-water communication are restricted by the nature of the environment. The complex, time-varying nature of the environment exclude certain modulation schemes due to amplitude and phase fluctuations, mostly caused by multipath signals.

The faded multipath channel creates a significant problem when designing a point-to-point communication system, especially when the utilisation of directional transducers and receiver arrays cannot be used due to the nature of a positioning system. There are prescribed conditions in which the system has to work. The system has to have no directionality because the transponders (i.e. Seabed, Diver & AUV) can be initiated at any position. Hence, multiple element receivers and spatial diversity processing algorithms cannot be used to overcome problems associated with amplitude and phase fluctuations caused by faded multipath interference. The cost, physical size and complexity of such arrays would not the feasible for a positioning system discussed here. The modulation scheme developed for this research is for transmission and reception via a single omni-directional transducer.

The various type of modulation; Amplitude, Phase and Frequency can be generalised and shown in the form:

$$s(t) = A\cos(\varpi_c t + \phi) \qquad (5.2)$$

If the information signal is $x(t)$, then:

For Amplitude Modulation (AM): $A = A_0 + K_A x(t)$ .... (5.3)

For Phase Modulation (PM): $\phi = \phi_0 + K_P x(t)$ .... (5.4)

For Frequency Modulation (FM): $\frac{d\phi}{dt} = K_P x(t)$ .... (5.5)

5.2.1 On–Off Keying (OOK) or Amplitude Shift Keying (ASK)

This technique is probably the simplest conceptually, comprising a sequence of binary pulses, in which the 1's turn on the carrier frequency of amplitude A, and the 0's turn off the carrier frequency. The spectrum of the OOK signal depends on the serial binary data sequence being transmitted.[5.26] OOK is the simplest form of Amplitude Modulation (AM) and is often termed Amplitude Shift Keying (ASK). ASK is used in underwater communications but, due to the bandwidth limiting effects of the hydrophone and the fading multipath environment this technique is susceptible to noise and inter-symbol interference. However, the susceptibility to noise and ISI is occasionally exploited to evaluate the effectiveness of error correcting codes.[5.34] In analogue data transmission a carrier signal is Amplitude Modulated by a lower frequency data signal, hence the transmitted information is superimposed as the envelope of the carrier signal. Demodulation is performed by removing the carrier frequency using a low-pass filter, leaving the baseband analogue data signal. In digital terms, ASK can be extended from simple OOK by varying the signal amplitude level to represent a different binary sequence. For digital signals ASK is often combined with Phase Shift Keying (PSK) to give multi-symbol signaling schemes. The resultant signals are called Quadrature Amplitude Modulation (QAM) signals. These signals may be interpreted as having multilevel amplitude modulation applied independently on each of the two quadrature carriers. The demodulator incorporates level detectors on the outputs of a Quadrature Phase Shift Keying (QPSK) demodulator to recover the QAM encoded data.

5.2.2 Frequency Shift Keying (FSK)

The FSK modulation technique in its basic form is such that one signal frequency represents a data bit state 1 and another signal frequency represents a data bit state 0. Therefore the bandwidth requirement for frequency signalling modulation schemes is higher than for phase and amplitude signalling schemes. FSK modulation techniques are often used in noisy channels and at low data rates, because as the data rate increases so does the bandwidth requirements. The relatively large spectral side lobes caused by the phase discontinuities at the frequency change instants can often disrupt the reception of the data at the receiver. A FSK technique that eliminates unwanted high frequency components caused at the switching instant is known as phase FSK or Minimum Shift Keying (MSK). MSK involves transmitting a FSK signal with continuous phase transition between bit periods. In practice, depending on the operating frequency, basic FSK is adequate as the high frequency components of the signal generated by switching frequency is filtered out by the medium as well as the projector and receiver element. An advantage of FSK is the simplicity in which the symbol alphabet (tones) can be generated by using an oscillator. To produce continuous phase transmission a single oscillator can be divided down to the required symbol frequencies. The FSK receiver has to make a decision as to which frequency has been received over the symbol period, hence demodulation of the modulated received signal. The demodulation of the FSK signal can be coherent or non-coherent, as the data is carried in the frequency of the signal and not the phase.

5.2.3 Multiple Frequency Shift Keying (MFSK)

Multiple frequency shift keying technique is similar to FSK, but instead of two frequencies that represent single bits of data, multiple frequencies are used to represent multiple bits. In general terms the technique involves an arbitrary number of *M* frequencies (at least two) that represent the transmitted data. Each frequency tone is transmitted separately, not in parallel as with Space Frequency Shift Keying (SFSK). [5.25] MFSK performance is generally characterised by a rapid increase in performance for an increase in *M* frequencies but then becomes negligible for large *M*.

Data Bits	**FSK**		**4FSK**
	Symbol Period (T)	Symbol Period (T)	Symbol Period (T)
00	F_1	F_1	F_1
01	F_1	F_2	F_2
10	F_2	F_1	F_3
11	F_2	F_2	F_4

Table 5.1Assignment of two bits of data

Table 5.1 shows the increase in data throughput that can be achieved by increasing the number of frequencies *M*, while maintaining a constant symbol period. As can be seen, the data throughput for a symbol period increases twofold and so has the number of frequencies. Increasing the number of frequencies further does not give the same increase in data throughput, e.g. 8FSK = 3 bits/T, 16FSK = 4 bits/T and so on. Hence, the number of frequencies M to represent D discreet data bits is:

$$M = 2^D \qquad (5.6)$$

Reference[5.35] outlines application where MFSK techniques have been used and present typically performance figures.

To increase the reliability, frequency hopping multi-frequency shift keying modulation techniques are often employed. For example; a system has a bandwidth of 15KHz and is divided into 16 frequency bins, each with a 1KHz bandwidth. Assume binary FSK, there are two valid frequencies per symbol. Following the transmission of the symbol the valid frequencies change to 2 of now 14 available frequencies, hence the frequency hops. Once all of the 16 frequencies have been transmitted the process is repeated, according to hopping sequence. This is a simple example of frequency hopping, schemes that are used in the underwater environment are often more complex. [5.14][5.27][5.33] The reliability of a frequency hopping system is maintainable as long as the channel reverberation is not too long. The period for the system to return to the same band should be longer than the channel reverberation time T_r. Equation (5.7) can be used to estimate the maximum reverberation time to achieve sustained reliability.

$$T_r \approx (n-1)T_{bd} \qquad (5.7)$$

Where T_{bd} is the baud rate duration and n is the specified number of frequency bands.

5.2.4 Minimum Shift Keying (MSK)

Minimum shift keying is also referred to as Continuous-Phase Frequency Shift Keying (CPFSK), hence no phase discontinuities occur during inter-bit switching. The phase information contained in a received signal is not fully exploited in coherent Binary FSK, other than providing receiver transmitter synchronisation. Using the phase information when performing detection, it is possible to improve the noise performance of the receiver significantly.[5.26] This improvement does, however, increase the complexity of the receiver. The phase is a continuous function of time, hence it is found that the modulated signal itself is also continuous at all times, including the inter-bit switching times. The phase of a CPFSK signal increases or decreases with time during each bit duration of *Tb* seconds.

To summarise, the desirable properties of the MSK signal are:

Constant envelope;

Relatively narrow bandwidth;

Coherent detection performance equivalent to that of QPSK.

5.2.5 Phase Shift Keying (PSK) and Differential PSK (DPSK)

Phase shift keying operates at one frequency, because the information (data) is conveyed in the phase component of the signal. The most basic form of PSK is Binary Phase Shift Keying (BPSK), which involves the assignment of a constant phase to each of the two possible bit values (1 or 0); the frequency and amplitude are kept constant. The symbols therefore consist of a constant frequency tone with one phase representing a bit one and a second phase representing a bit zero. In general, there is a lower bandwidth associated with PSK-modulation schemes than with FSK, and the bandwidth advantage is apparent when the modulation scheme utilises a large number of symbols. PSK modulation is particularly effective in higher-rate communication channels where phase noise and distortion is not a problem. BPSK gives the minimum Bit Error Rate (BER) due to the decision threshold being along the zero crossing of the waveform. For an error to occur, noise *n* has to be greater than *A*, the peak amplitude of the signal, as shown in Figure 5.6 .

Differential phase-shift keying (DPSK) can be considered as a non-coherent version of PSK. It eliminates the need for a coherent reference signal at the receiver by combining two basic operations at the transmitter: differential encoding of the input binary pattern and phase shift keying. The receiver measures the relative phase difference between the waveforms received during two successive bit intervals. For example; the transmission of a symbol 1 followed by a symbol 1 would require no phase change as the state has not changed, whereas if the first symbol is followed by a symbol 0, this would require a phase change to indicate the bit change. DPSK requires a reference bit at the start of the communication to denote the differentially encoded sequence.

5.2.6 Offset QPSK (OQPSK)

OQPSK is QPSK with an offset in the alignment of the two-baseband waveforms. In standard QPSK, the odd and even pulse streams are both transmitted at a rate of ½T bits/s and are synchronously aligned, such that there transmissions coincide, as shown in Figure 5.3 A frequent mis-conception with OQPSK is that a 180° phase change cannot occur in a symbol period; this is not true. However, a 180° phase change does not occur at any one instance. Instead 90° phase changes can occur every data bit period *T*, hence in a symbol period *2T*, two 90° phase changes can have occurred.

In OQPSK the data pulse streams $d_I(t)$ and $d_Q(t)$ are staggered and thus do not change states simultaneously. The possibility of the carrier experiencing a 180° instantaneous phase change is eliminated, since only one component can make a transition at an one time. Changes are limited to 0° and ±90° every *T* seconds. If a QPSK modulated signal undergoes filtering to reduce the spectral side-lobes, the resulting waveform will no longer have a constant envelope and in fact at the 180° phase transition the envelope will momentarily go to zero. When OQPSK undergoes band limiting, the resulting ISI causes the envelope to droop slightly in the region of the ±90° phase transitions, but since the 180° phase transitions have been avoided the envelope will not go to zero.[5.22]

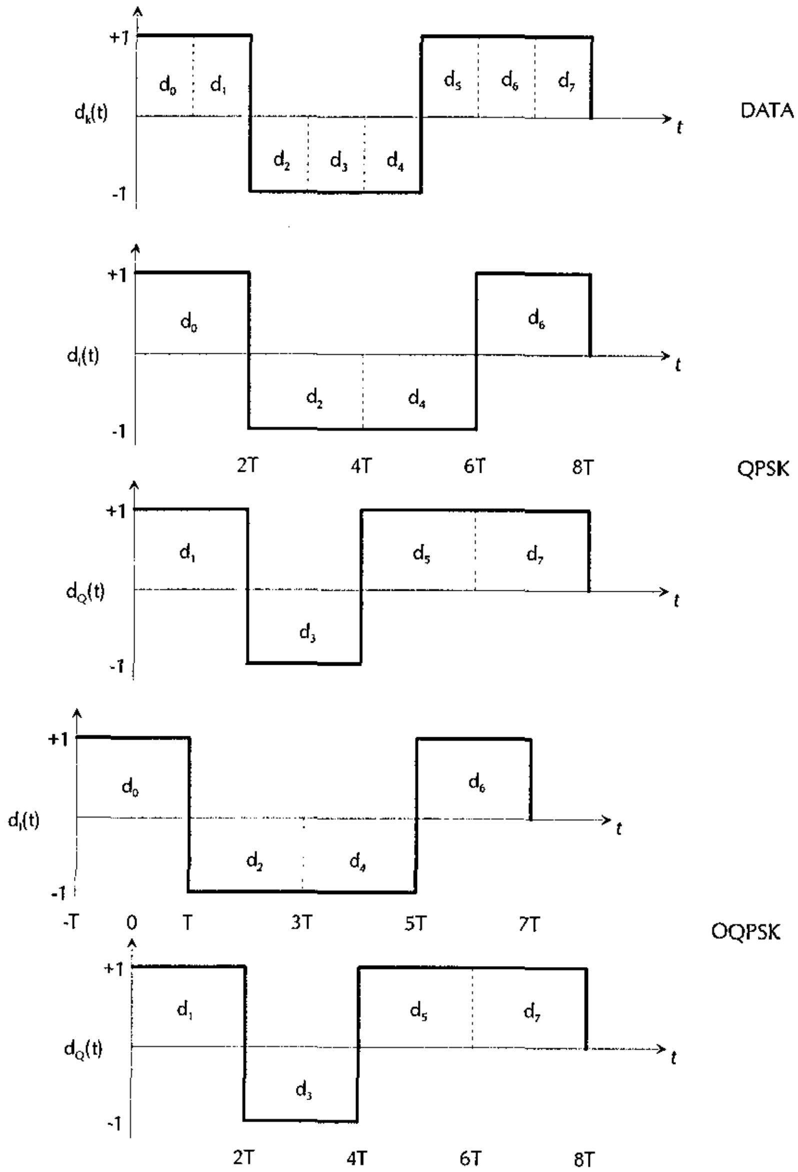


Figure 5.3 QPSK and Offset QPSK encoding

5.2.7 Multiple Phase Shift Keying (MPSK)

MPSK increases the data rate of straight PSK by encoding data into *M* phase positions. Quadrature PSK (QPSK) coveys twice as much data per symbol as BPSK. The valid phasor position for QPSK are 0°, 90°, 180° and 270° representing two bits of data (i.e. 00, 01, 11, 10) as shown in Figure 5.3 . This can be increased further to 8 phases, 16 phases and so on. As with MFSK the initial gain in data rate is high and the number of data bits per symbol is calculated using Equation (5.6). As the number of phase positions increases, the probability of channel noise, causing the data to be decoded incorrectly, also increases. Hence, the BER increases for the same transmitter and receiver characteristics, with respect to the number of bit per symbol.

QPSK has the same bit error rate probability as BPSK for the same energy per bit, Eb/No. The QPSK bit stream can be partitioned into an even and odd (In phase *I* and quadrature Q) stream; each stream modulates an orthogonal component of the carrier at half the bit rate of the complex stream. The natural orthogonality of the 90° phase shifts between adjacent QPSK symbols results in the bit error rate probabilities being equal for both BPSK and QPSK signalling. It is important to note that the symbol error probabilities are not equal for BPSK and QPSK signalling.[5.22] pp. 223

5.2.8 Pulse Position Modulation (PPM)

This data transmission scheme conveys data to receiving units not through data encoded into the signal transmission, but by the position of the signal transmission. In PPM, the receiver can be synchronised to the transmitter so that absolute timings are decoded or, in non-synchronous system, incremental timings are decoded. In the absolute or synchronous system a start bit is required to synchronise the two systems and provide a timing datum. In a synchronous system a re-synchronisation pulse may be required, ensuring that timing drift does not cause data decoding errors. A non-synchronous or incremental timing system can suffer similar problems, as the timing information for a whole data packet can be in error due to one bit error at the start of the data packet. With both systems there is a need for timing datum to be transmitted periodically to ensure data integration.

PPM techniques have been successfully implemented for underwater communications using both acoustics and optical links.[5.38][5.39] PPM can be combined with FSK or PSK to increase the data rate of a PPM communication link. The PPM combined with FSK approach is used by most of the commercially available acoustic positioning systems. A combination of PPM and PSK is discussed later in this chapter.

5.2.9 Modulation Techniques Summarised

Various modulation schemes have been discussed; however certain techniques are more appropriate for underwater communications. Furthermore, certain schemes are more appropriate to the research and constraints imposed by the positioning system. Comparisons of the various modulation techniques discussed in this chapter are shown in Table 5.2 , together with comments on the suitability of the scheme in the context of underwater communication.

Modulation Scheme	Bandwidth Efficiency	Anti-fading Properties	Anti-ISI Properties	Comments
OOK or ASK	Poor	Poor	Poor	Easy to implement
FSK	Moderate	Moderate	Moderate	Easy to implement
MFSK	Poor	Moderate	Good	Difficult to implement, can often require DSP to perform FFTs.
MSK or CPFSK	Moderate	Moderate	Moderate	Relatively complex modulator and demodulator, does give QPSK BER performance.
MPSK	Good	Moderate	Moderate	Carrier is required, easy to implement
DPSK	Good	Moderate	Moderate	No carrier recovery is required, easy to implement
QAM	Good	Poor	Poor	Difficult to implement, adaptive equalisation needs to employed
PPM	Poor/ Moderate	Moderate	Poor/ Moderate	Can be good if combined with FSK

Table 5.2 Comparison of various modulation methods

5.3 Noise and Error Probability

The maximum advantage over noise in a transmission medium is obtained by using a binary code, because a binary symbol withstands a relatively high level of noise and is easy to regenerate.[5.26] Pulse-Coded Modulation (PCM) systems are considerably more complex than Pulse Amplitude Modulation (PAM), Pulse-Duration Modulation (PDM) and Pulse Position Modulation (PPM) system in that the signal data is subjected to a greater number of operations. The essential operations in a PCM system are sampling, quantisation, encoding and decoding. Sampling of the received signal using an ADC generates a digitised waveform of the points at the sampling instants. The sampling frequency must be at least twice the expected signal frequency at the receiver to ensure that correct signal regeneration can be achieved. Quantisation produces discrete digital steps that represent the received analogue signal. The higher the resolution, the smaller the quantisation steps, and hence a closer representation of the received waveform. The quantising errors consist of the difference between the received and transmitted signals of the quantiser. The effect of noise is to introduce bit errors into the received PCM wave, with the result that, in a binary system, a symbol 1 occasionally is mistaken for a symbol 0, or vice versa. The measurement of the error rate or probability of error in a PCM system indicates the fidelity of the information transmission in the presence of noise. Consider a binary-encoded PCM wave *s(t)* consisting of a sequence of binary digits, in which a symbols 0 and 1 are represented by the level zero and *A* volts, respectively. This type of encoding represents OOK. The received signal *x(t),* consists of the PCM wave *s(t)* plus white Gaussian noise, *n(t),* (zero mean $\mu = 0$) with a power spectral density of $N_o/2$. If the received signal is pre-processed by a filter of bandwidth *B* that is large enough to pass the PCM wave essentially unchanged and yet small enough to limit the effects of noise *w(t),* the filtered received version of the waveform is expressed as *y(t),* as follows

$$y(t) = s(t) + n(t) \qquad (5.8)$$

where the noise *n(t)* has zero mean and variance $\sigma^2 = N_oB$.

In determining the state of each data bit, either 0 or 1, the filtered signal *y(t)* is sampled every T_b seconds, where T_b is the bit period. The value of each sample is then compared with some predetermined threshold. In a system with equal likelihood of a 0 or 1; the threshold levels to give maximum reliability are chosen to be mid-way between the bit levels. The Bit Error Rate (BER) is the probability *P* of a single bit being corrupted in a set number of bits transferred. Thus a BER of 10^{-4} means that, on average, 1 bit in 10,000 will be corrupt.

The expressions for the BER for coherent binary PSK, conventional coherent binary FSK with on-bit decoding, DPSK, non-coherent binary FSK, coherent QPSK, and coherent MSK, when operating over an Additive White Gaussian Noise (AWGN) channel is tabulated in Table 5.3 [5.26]

Signaling Scheme	**Bit Error Rate**
Coherent binary PSK	
Coherent QPSK	½ $\mathrm{erfc}(E_b/N_o)^{-2}$
Coherent MSK	
Coherent binary FSK	½ $\mathrm{erfc}(E_b/2N_o)^{-2}$
DPSK	½ $\exp(E_b/N_o)$
Non-coherent binary FSK	½ $\exp(-E_b/2N_o)$

Table 5.3 BER formulae for digital modulation schemes

5.3.1 Error Control[5.28]

When a data packet is sent from transponder A to transponder B, there is a requirement to know whether it has arrived without error. The simplest form of error check is to attach a parity bit to the data packet or each data byte. The parity bit is chosen so as to make the total number of one-bits either always even (even parity) or always odd (odd parity). Any single bit error in a data packet or data byte will thereby be detected. When errors are sufficiently rare, and do not occur closely bunched in time, the use of parity provides sufficient error detection.

The two factors that determine the type of error detection scheme used are the bit error rate (BER) of the communication system and the type of errors. The errors can occur as random single-bit errors or as groups of contiguous strings of errors, often referred to as burst errors. In real situations and especially underwater communications, a single noise event is likely to disrupt more than one bit. Since the parity bit has two possible values (0 or 1), it gives only a 50% chance of detecting an erroneous data byte with more than one wrong bit. The 50% probability of detecting a corrupt data packet is, in most applications, not good enough. Most communication protocols use a multi-bit generalisation of the parity bit called a Cyclic Redundancy Check (CRC). In typical communication applications a 16-bit CRC is implemented, and the possibility of a random error going undetected is 1 in 2^{16} (or 1 in 65536). Moreover, *M*-bit CRCs have the mathematical property of detecting all errors that occur in *M* or fewer consecutive bits, for any length of message.[5.29] Since noise in underwater communication channels tends to be time varying and in bursts, with short sequences of adjacent data bits becoming corrupt, the consecutive-bit property of the CRC is highly desirable.

The error detection can be further improved by not only being able to detect that an error has occurred but to be able to correct the error at the receiver. The receiver can correct corrupt data bits due to the extra error correction code encoded into the signal before transmission. Some of the other main types of error control are discussed briefly below; however these are not elaborated further as full explanations of the various types of error control is well documented.

Convolutional Codes (RW)

In block coding, the encoder accepts a *k*-bit message block and generates an *n*-bit code word. Thus, code words are produced on a block-by-block basis. Clearly, provision must be made in the encoder to buffer an entire message block before generating the associated code word. There are applications, however, where the message bits come in serially rather than in large blocks, in which case the use of a buffer may be undesirable. In such situations, the use of convolutional coding may be the preferred method. A convolutional coder generates redundant bits by using modulo-2 convolutions, hence the name.

The encoder of a binary convolutional code with rate *1/n*, measured in bits per symbol, may be viewed as a finite state machine that consists of an M-stage shift register with prescribed connections to n modulo-2 adders, and a multiplexer that serialises the outputs of the adders. An *L*-bit message sequence produces a coded output sequence of length *n(L+M)* bits. The code rate is therefore given by

$$r = \frac{L}{n(L+M)} \text{ bits/symbol}$$

Typically, we have L>>M. Hence, the code rate simplifies to

$$r \approx \frac{1}{n} \text{ bits/symbol}$$

The constraint length of a convolution code, expressed in terms of message bits, is defined as the number of shifts over which a single message bit can influence the encoder output. In an encoder with an M-stage shift register, the memory of the encoder equals *M* message bits, and $K = M + 1$ shifts are required for a message bit to enter the shift register and finally come out. Hence, the constraint length of the encoder is *K*.

The system captures a complete data packet and then post-processes the data due to the high sampling rate in which the data is captured (see chapter 3). As discussed, convolutional codes are generally used in situation where the data is not buffered, which not the case with the positioning system. Therefore, an error detection code that is generated on a block of data is more appropriate for the system.

Linear Block Codes

A code is said to be linear if any two code words in the code can be added in modulo-2 arithmetic to produce a third code word in the code. Consider a (n,k) linear block code, in which k bits of the n code bits are always identical to the message sequence to be transmitted. The n - k bits in the remaining portion are computed from the message bits in accordance with a prescribed encoding rule that determines the mathematical structure of the code. Accordingly, these n - k bits are referred to simply as parity bits. Block codes in which the message bits are transmitted in unaltered form are called systematic codes. For applications requiring both error detection and error correction, the use of systematic block codes simplifies implementation of the decoder.

BCH Codes

The Bose Chaudhuri Hocquenghen (BCH) codes are a generalisation of Hamming codes, which allow multiple error correction. These codes are easily defined in terms of the roots of the generator-polynomial. Subdivisions of these polynomial generated codes are Reed Solomon, generalised Reed Mueller, projective geometry codes, Euchdean geometry codes and quadratic residue codes. Each of these classes of codes is described by a specific algorithm for constructing the code. The classes form overlapping sets so that a particular code may be a BCH code and also a residue code or it may be a generalised Reed-Mueller code and also a BCH code, etc. Polynomial-generated codes are important for several reasons. Firstly, they are relatively easy to implement in hardware as a simple feedback shift register. Secondly, this class contains codes whose minimum distance is close to the best that can be found, especially for block lengths of the order of 100 or less. There also exists several decoding algorithms that enable moderate amounts of hardware to decode the codes.

Coding Gain

Communication system performance is the ratio of energy per information symbol to noise spectral density E_b/N_o that is required to achieve a given probability of error. The use of error correcting codes is often termed as coding gain and describes the improvement that is achieved when a particular coding scheme is implemented. To determine the coding gain, the probability of error versus E_b/N_o of a coded and uncoded signal can be plotted on a graph and a comparison drawn. For example[5.28], a (23,12) Golay code with hard-decision decoding gives a coding gain of 2.15dB at $P_e = 10^{-5}$ and 1.35dB at $P_e = 10^{-3}$. At very low signal-to-noise ratios, the coding gain can actually become negative. This threshold phenomenon is common to all coding schemes. There will always exist a signal-to-noise at which the code loses its effectiveness and actually makes the situation worse.

Interleaving

Most Forward Error Correction (FEC) schemes are designed to combat random independent errors. Such errors occur in memoryless propagation channels (i.e. no multipath present - Gausian). When the channel is not memory less, signals arrive at the receiver via a number of propagation paths. The net effect is a received signal that is the vector sum of all of these multipath signals. Since all propagation paths are time variant, resultant signal strength and phase are also time variant. This varying signal (which can only be described statistically) is termed a fading signal. Such signals are prone to produce burst errors in the receiver decoder. Interleaving is used to alleviate these burst error effects by introducing time diversity into the transmitted data. Sufficiently deep interleaving can make the propagation channel appear memoryless. As a result, less powerful FEC codes can be used to correct the 'burst' error effects. In general, an increase in the interleaving period results in improved decoder error rate performance. It is prudent to implement interleaving when moderately long messages are transmitted. Although there is negligible increase in processing demand, additional memory is required at the transmitter and receiver. In a high data rate mobile communication link operating at Mbytes/s the decoding delay introduced by the interleaving process may be unacceptable. However, for the data rates considered in acoustic communications (KB/sec) the delay is generally acceptable. The incorrect choice of interleaving or incorrect implementation of a scheme may completely negate the effectiveness of the FEC processing. Poor choice of an interleaving scheme may result in a decoder error rate that degrades by the application of FEC coding.

The principles of interleaving fall into four categories:

- Diagonal
- Block
- Inter-block
- Convolution

A summary of the advantages and disadvantages of the above interleaving schemes is shown in Table 5.4 Interleaving schemes.

Cyclic Codes

Cyclic codes form a subclass of linear block codes. Indeed, many of the important linear block codes discovered to date are either cyclic codes or closely related to cyclic codes. An advantage of cyclic codes over most other types of codes is that they are easy to encode. Furthermore, cyclic codes posses a well-defined mathematical structure, which has led to the development of very efficient decoding schemes for them.

A binary code is said to be cyclic code if it exhibits two fundamental properties: (i) Linearity property: the sum of any two code words in the code is also a code word, (ii) Cyclic property: the cyclic shift of a code word in the code is also a code word.

Interleaving Scheme	**End-to-end delay in symbols**	**Required Storage in symbols**	**Comments**
Diagonal	6n m bits/symbol n symbols	3n	short delay Only disperses burst errors over 2 adjacent blocks
Block	2WD-2W+2 Fill matrix serially horizontally read out vertically W matrix columns D matrix rows	WD	D must be > burst error length Not robust for periodic sequences of single burst error spaced d symbols apart
Inter-block	B^2N N symbols dispersed in B blocks		Combats periodic interference Possibly more commercially secure Constraint placed on block size Long interleave delay Transmission expanded by (B-1) blocks Introduces dead time
Convolution	W(D-1) W=DM M = memory in symbols	W(D-1)/2	Similar performance to Block interleaving but requires less memory.

Table 5.4 Interleaving schemes

Cyclic Redundancy Check (CRC)

Mathematics underlying CRCs is a polynomial over the integers modulo 2. Any binary message can be represented by a polynomial expression with coefficients of 0 and 1, e.g. the data packet 10110101 is the polynomial $x^7 + x^5 + x^4 + x^2 + 1$. Since 0 and 1 are the only modulo 2 integers, a power of x in the polynomial is either present (1) or absent (0). An *M*-bit long CRC is based on a particular primitive polynomial of degree *M*, called the generator polynomial. The choice of which primitive polynomial to use is only a matter of convention. To keep the data packet as short as possible an 8-bit CRC has been implemented, hence giving a 1 in 2^8 (256) chance of a random error going undetected. The polynomial for an 8-bit CRC is $x^8 + x^5 + x^4 + 1$ and is illustrated in Figure 5.4 [5.30]

It was considered during the development of the protocols that error detection would be sufficient for the positioning system. The overhead of implementing a Reed Solomon error detection and correction algorithm such as a RS(15,7,4) more than doubles the size of the packet. A RS(15,7,4) code has a block length of 15 symbols, 7 of which represent the required information content of the block; the remaining symbols are used for parity checking, where each 1 symbol comprises 4 bits. Also, due to the protocol design and use of dynamic packet lengths, there would be further redundancy when transmitting short commands. However, if the system was required to communicate medium amounts of data in a noisy environment then implementation of a RS or Golay code could improve the systems performance.

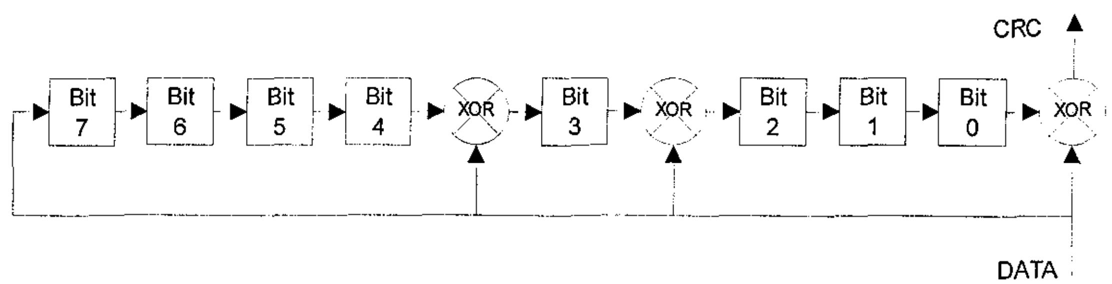


Figure 5.4 Cyclic Redundancy Check

The cyclic redundancy check is simple to implement in software and the encoding and decoding time is minimal. The eight-bit CRC was deemed sufficient, as the maximum packet length including the CRC is 69 bits. Currently the CRC accounts for between 11.5% and 50% of the data packet transferred. On the shorter packets, this becomes a significant overhead.

5.4 Correlation and Synchronisation

There is often a need in signal processing to quantify the degree of interdependence of one process upon another, or to establish the similarity between one set of data and another.[5.31] Applications of correlation are found in sonar systems for range and position finding in which transmitted and reflected waveforms are compared, in detection and identification of signals in noise, in the computation of the average power in waveforms and in many other fields. Consider the correlation of two simultaneously sampled waveforms that vary similarly point for point, then a measure of their correlation can be obtained by taking the sum of the products of the corresponding pairs of points. Considering the case of two independent and random data sequences can show the attenuating effects of correlation on noise. In this case the sum of the products will tend to a decreasingly small random number as the number of pairs of points is increased. This is because with a random sequence of data the probability of positive and negative numbers occurring is equally likely, so the product of pairs of points tends to be self cancelling on summation; hence a mean of zero when considering an infinite number of points. However, in a finite data sequence there exists a degree of correlation. The cross-correlation of two data sequences of N number of samples can be expressed mathematically as:

$$r_{12} = \sum_{n=0}^{N-1} x_1(n) x_2(n) \qquad (5.9)$$

where $r_{12}(n)$ is the cross-correlation of the two data sequences $x_1(n)$ and $x_2(n)$.

The result of Equation 5.9 is dependent on the number of sample points N; this can be corrected for by normalising the result to the number of points by dividing by N, therefore averaging the sum of the products.

The above definition is not particularly useful as in some cases it can indicate zero correlation although the two waveforms are 100% correlated. This may occur when the two waveforms are out of phase, which in signal processing will often be the case. To overcome phase differences it is necessary to shift one of the waveforms with respect to the other. Thus the formula for cross-correlation becomes:

$$r_{12}(j) = \frac{1}{N}\sum_{n=0}^{N-1} x_1(n)x_2(n+j) \quad \text{(5.10)}$$

where j represents the amount of lag which is the number of sampling points by which x_2 has been shifted.

Often when correlating two waveforms their phase relationship will not be known so the correlation is computed for a number of different lags in order to establish the largest value of the correlation, which is then taken to be the correct value.

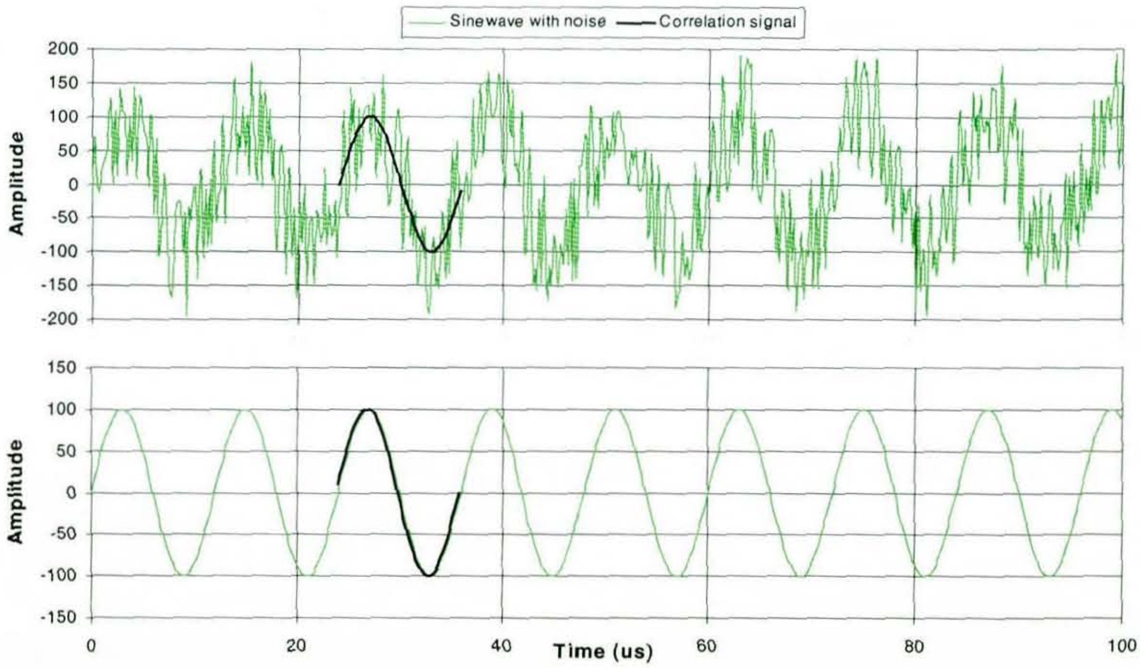


Figure 5.5 Synchronisation precision using correlation

The packet nature of data transmission between transponders requires the receiver to be able to synchronise to the captured packet, so that both coherent transfer of data and accurate time-of-flight measurements can be made.

There are two basic modes of synchronisation: when coherent detection is used, knowledge of both the frequency and phase of the carrier is necessary (*carrier synchronisation*). To perform demodulation, the receiver has to know the instants of time at which the modulation can change its state. That is, it has to know the starting and finishing times of the individual symbols, so that it may determine when to sample the received signal (*symbol synchronisation*).

These two modes of synchronisation can be coincident with each other, or they can occur sequentially. Naturally, in a non-coherent system, carrier synchronisation is of no concern.

There are two synchronisation schemes; data-aided and non-data aided. In non-data aided synchronisation the receiver has the task of establishing the synchronisation by extracting the necessary information from the modulated signal. Both throughput and power efficiency are thereby improved but at the expense of an increase in the time taken to establish synchronisation. In data-aided synchronisation systems, a preamble is transmitted along with the data-bearing signal in a time-multiplexed manner on a periodic basis. The preamble contains information about the carrier and symbol timing, which is extracted by processing the receiver output. Its limitations are two-fold: (1) reduced data throughput efficiency that is incurred by assigning a certain proportion of each transmitted frame to the preamble, and (2) reduced power efficiency by allocating a certain fraction of the transmitted power to the transmission of the preamble.

At the start of the data packet there are several synchronisation cycles of a known frequency to allow accurate phase recover of the received packet. Figure 5.5 shows the phase recovery precision of synchronisation cycles using cross-correlation, even when the signal is distorted by noise. The successful and accurate phase recovery of the received data packet determines whether the data can be correctly decoded. Also, the accuracy of the phase correlation determines the timing accuracy as the start-bit that represents the time datum is detected by looking for the start bit (phase change) or sequence of data bits in the received data. Figure 5.14 shows a typical data packet and highlights the synchronisation cycles, start-bit and individual data bits.

In high noise environments correlating a single cycle is often not sufficient to obtain good symbol lock onto the synchronisation pulse. In this situation part of the synchronisation cycle and start bit are replaced with a *gold code*[11], which is a sequence of bits that gives an unambiguous correlation peak when the waveforms are aligned. Some codes that possess these properties are Barker and Willard codes. Barker codes (11-bit) are used for the Direct Sequence (DS) spreading function in 2.4GHz wireless LANs. This sequence is well known in the industry as having *optimal* autocorrelation properties. It produces a single peak and uniformly low sidelobes when correlated against versions of itself. Thus, it has very good rejection of multipath. The autocorrelation properties of Barker words are often measured with all-zero data appended to either side of the sequence being correlated. A problem relating to this is that the data following the *gold code* could mimic the same sequence, hence causing a correlation peak of similar amplitude in the data stream itself. The choice of *gold codes* is limited especially when only considering 3 to 8 bit codes. Three to thirteen bit Barker and Willard codes are listed in Table 5.5 , and apart from the four bit Willard code they are all asymmetric.

N	Barker Sequences	Willard Sequences
3	110	110
4	1110 or 1101	1100
5	11101	11010
7	1110010	1110100
11	111000010010	11101101000
13	1111100110101	1111100101000

Table 5.5 Barker and Willard code sequence

The start-bit detection probability is improved by correlating over a greater number of cycles; however the drawback is that the correlation procedure is computationally intensive. For a single cycle cross-correlation (5MHz sampling, 60 samples per cycle) there are 360 multiples and a similar number of additions. If a cross-correlation peak is computed for the whole of the synchronisation pulse it takes approximately 50ms. The processor performs 16-bit multiplications and 32-bit additions, which takes a considerable number of clock cycles. A DSP would be capable of performing a

[11] Gold codes are specific sequences of psuedo-random numbers that can be generated using two feedback shift register and posses the following properties: (i) Contains approximately the same number of ones and zeros, (ii) be approximately orthogonal to there codes, (iii) be approximately orthogonal to themselves when delay/shifted, (iv) be easy to generate.

correlation task in a fraction of the time as they have dedicated registers that hold and can add 32-bit numbers in a single instruction.

However, by losing some resolution in the captured data and performing a 30-point cross-correlation instead of a 60-point correlation the correlation period can be reduced to a usable time frame. The maximum packet processing time is the deadtime following a packet capture, which is nominally 200ms. If the packet-decoding period exceeds the deadtime, the transponder is not able to respond to a command that has not yet been decoded. There is also a risk of overloading the transponder with received packets, which inevitably would slow down the system operation. So to ensure that the operating speed of the system is a constraint of the medium rather that the processing power, the maximum packet-decoding period was set to 100ms.

5.5 Modulation Schemes

The requirement of the system is to be able to exchange relatively small amounts of data between transponders in a packet transmission form. The hardware has been designed to facilitate various communication techniques; however due to the packet nature of the communication link and the environment there is a need for the packets to be as short as possible. As discussed earlier, this presents a significant problem due to the omni-directional transmission of acoustic data and the electro-acoustic properties a spherical transducer (hydrophone/projector combined). The Q of the hydrophone is around 6, hence implementing a FSK communication technique would require at least 6 cycles/bit (at the resonant frequency of the hydrophone this corresponds to 72μs/bit). Assuming 8 cycles proved sufficient without significant loss in performance, the packet size would still be large. For example; 8 cycles/bit = 96μs/bit, the average packet length being 26bits - 2.5ms, before any framing bits and synchronisation cycles have been included.

The packet transmission is required to enable multiple transponders to have access to the medium under the control of the master. The transponders are capable of transmitting and *time stamping* the arrival of data packets to an extremely high level of accuracy. The accurate time logging enables the transponder array to calculate the precise separation of transponder to within a centimetre. The timing precision of the system is discussed in detail in Chapter 6. However, this precise timing could also be used for a PPM scheme due to the high time-of-arrival accuracy.

5.5.1 Modulation Scheme 1

The modulation scheme 1 was developed to represent one data bit per two signal cycles, hence one symbol period *Ts* represent one bit of data, by implementing a coherent BPSK modulation scheme. Data is transmitted as discrete symbols *S*, hence a binary transmission, gives:

$$S \in \{S_1, S_2\} \qquad (5.11)$$

Initially a conventional BPSK modulation scheme was devised and programmed. Although simulations of the scheme indicated that it would work well; acoustic tests in the tank indicated otherwise. The scheme totally failed and never succeeded in transmitting a single packet of data without errors. This was due to the characteristics of the hydrophone and channel when attempting to transmit signals with 180° instantaneous phase changes.

To resolve this problem, one can either; increase the symbol period, or reduce the instantaneous phase change. Increasing the symbol period will reduce the data rate, as the data rate is directly proportional to the symbol rate (Binary PSK, hence one symbol represents one bit of data). Reducing the phase change from 180° to 90°, maintains the symbol rate, but increase the probability of error.

The modulation scheme shown in Figure 5.6 is normal QPSK, hence there are four possible phase positions S_1, S_2, S_3 and S_4 where A is the peak amplitude of the signal. However, as indicated in Figure 5.6 the phasor can only rotate clockwise and by a maximum of 90° per symbol. The modulation scheme is BPSK as there are only two valid symbols at any one time, but instead of 180° phase shift to indicate a bit change a 90° phase change is used. Restricting the maximum phase change reduces the bandwidth, hence reducing the hydrophone/projector recovery time following a phase change. This is similar to Offset QPSK, were the maximum phase change is restricted to 90° for the same reasons, but in convention OQPSK the phasor can rotate in either direction. The hydrophone responded better to phase retardation, rather than phase advance because the carrier frequency was set to 83.3KHz, hence above the natural resonance of the hydrophone, which is at 80KHz.

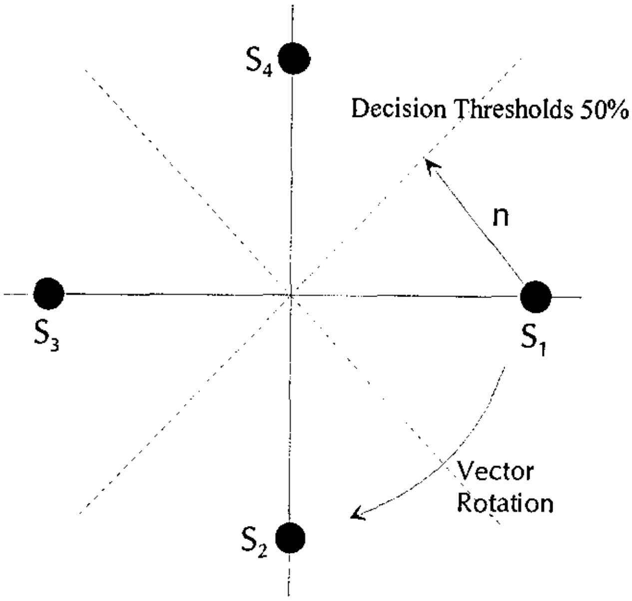


Figure 5.6 Decision Thresholds

The receiver recovers the phase of the transmitted signal by synchronising onto the packet header cycles, hence a coherent PSK communication link. The symbols and hence as it is a Binary modulation scheme the data, is differential encoded. That is a phase change is caused by a 0 to a 1 transition or vice versa in the data being encoded.

Throughout all the following derivations, it is assumed that each of the possible symbols is equally likely. The probability of an error is therefore evaluated as follows:

$$P_e = P\left\{A + n_c < \frac{A}{2}\right\} = P\left\{-A + n_c > \frac{-A}{2}\right\} \quad (5.12)$$

with n_c Gaussian and $\mu = 0$ (zero mean)

$$P_e = P\left\{0 + n_c > \frac{A}{2}\right\} = P\left\{0 + n_c < \frac{A}{2}\right\} \quad (5.13)$$

$$P_e = P\left\{n_c > \frac{A}{2}\right\} \Rightarrow P\left\{n_c < \frac{-A}{2}\right\} \quad (5.14)$$

$$P_e = \frac{1}{2} - \frac{\int_{-\infty}^{-A/2} e^{-x^2/2\sigma^2} dx}{\int_{-\infty}^{\infty} e^{-x^2/2\sigma^2}} = \frac{1}{2} - \frac{1}{\sqrt{\pi}} \int_{-\infty}^{-A/2} e^{-x^2/2\sigma^2} dx \quad (5.15)$$

Integration by substitution is applied to get the expression in terms of the error function and subsequently the complementary error function.

$$P_e = \frac{1}{2}\left(1 - \frac{2}{\sqrt{\pi}} \int_{-\infty}^{-A/2} e^{-x^2/2\sigma^2} dx\right) \Rightarrow \frac{1}{2}\left(1 - erf\left(\frac{A}{2\sigma\sqrt{2}}\right)\right) \quad (5.16)$$

$$\therefore P_e = \frac{1}{2} erfc\left(\frac{A}{2\sigma\sqrt{2}}\right) \quad (5.17)$$

The probability of the encoded data being a 1 or 0 is:

$P_A = 0.5$

The probability of data bit 1 being decoded incorrectly due to noise is:

$$P_w = P\left\{n_c < \frac{-A}{2}\right\} \quad (5.18)$$

and the probability of decoding data bit 2 incorrectly is partly dependent on whether data bit 1 was decoded correctly.

Figure 5.6 indicates how a single bit error can corrupt the decoding of the whole packet of data. Consider symbol period 1, for which the phase options are S_1 or a transition to S_2. Assume there is no change in the data stream; hence the phase remains at S_1. However, noise n in the channel causes symbol period 1 to be decoded as S_2, therefore a change in phase. The reference level for symbol period 2 is now different from when the data was encoded, and due to the nature of encoding and decoding S_1 is no longer a valid phase. Generally, errors will be incurred following an error until a change in state occurs and the decoder *catches-up* with the data stream. So in effect the decoder can determine that a phase change did not occur due to the redundancy in the encoding of the data. This is similar to Trellis Code Modulation where there is redundancy in the encoding of the data. Often TCM is used to ensure maximum separation distance between adjacent symbols to reduce the error probability.

The probability of a symbol/data error occurring is:

$$P_e = P_w + P_A P_w$$

$$P_e = \frac{1}{2} erfc\left(\frac{A}{2\sigma\sqrt{2}}\right) + \frac{1}{4} erfc\left(\frac{A}{2\sigma\sqrt{2}}\right) \qquad (5.19)$$

The error function in equation (5.19) is graphed in Figure 5.9 along with several simulated conditions for the above modulation scheme.

5.5.2 Continuous Phase

The modulation scheme discussed above is similar to Offset QPSK, where the phase change is limited to a maximum ±90°. The main advantage of OQPSK is that it suppresses out-of-band interference, however there are still phase discontinuities. This was the motivation for the development of a Continuous Phase Modulation (CPM) scheme. CPM is also known as Minimum Shift Keying (MSK), which can be considered as a special case of Continuous-Phase Frequency Shift Keying (CPFSK), or a special case of OQPSK with symbol weighting. The symbol weighting for MSK is often sinusoidal and can be expressed in the CPFSK form as:

$$s(t) = \cos\left[2\pi\left(f_0 + \frac{d_k}{4T}\right)t + x_k\right] \qquad kT < t < (k+1)T \qquad (5.20)$$

Where f_0 is the carrier frequency, $d_k = \pm 1$ represents the bipolar data being transmitted at a rate $R = 1/T$, and x_k is a phase constant which is valid over the kth binary data interval. Notice that the tone spacing for MSK is one-half that of normal non-coherent FSK, hence the name; *minimum shift keying*.

The Continuous Phase modulation technique is considered as a special OQPSK, and is encoded and decoded as phase modulated carrier and not a frequency modulation. QPSK can be expressed in a quadrature representation, as shown in Equation (5.21)

$$s(t)=\frac{1}{\sqrt{2}}d_I(t)\cos\left(2\pi f_0 t+\frac{\pi}{4}\right)+\frac{1}{\sqrt{2}}d_Q(t)\sin\left(2\pi f_0 t+\frac{\pi}{4}\right) \quad (5.21)$$

OQPSK with sine weighting can also be expressed in a quadrature representation, as shown in Equation (5.22). The Inphase and Quadrature components are staggered as shown in Figure 5.3 and the weighting is so that at the phase transition (every *2T*) the amplitude is zero.

$$s(t)=d_I(t)\cos\frac{\pi t}{2T}\cos(2\pi f_0 t)+d_Q(t)\sin\frac{\pi t}{2T}\sin(2\pi f_0 t) \quad (5.22)$$

The symbol period for the modulation scheme discussed above is two cycles, *Ts* and the output clock rate is 1.25MHz; giving 15 samples per cycle at a carrier frequency of 83.3KHz. The maximum instantaneous phase change is 1° and the profile of the phase change with output samples is shown in Figure 5.7 The carrier frequency (83.3KHz) corresponds to a phase increment of 24° per output sample. For a phase change of -90° the average phase increment during the symbol period is 21° per output sample. During the first cycle the transition from the carrier phase increment of 24° to 21° is made, with a purposeful overshoot to force the hydrophone to the desired phase. The second cycle of the symbol is to allow the hydrophone, power-amplifier and pre-amplifier to stabalise before the sampling instant. As the data is differentially encoded, if the binary data changes another -90° phase change occurs. In this case, to retard the phase a further 90° the phase change per output sample remain unchanged, hence 21°. Hence a transmission of a digital data stream of 1010101010, will in fact be a tone at 73KHz. This can be considered as discussed earlier as CPFSK or as CPOPSK, where there are two frequencies F_1 and F_2. It is worth noting the chosen frequencies and where they lie on the hydrophone calibration, i.e. either side of the hydrophone natural resonant frequency, at points of similar sensitivities.

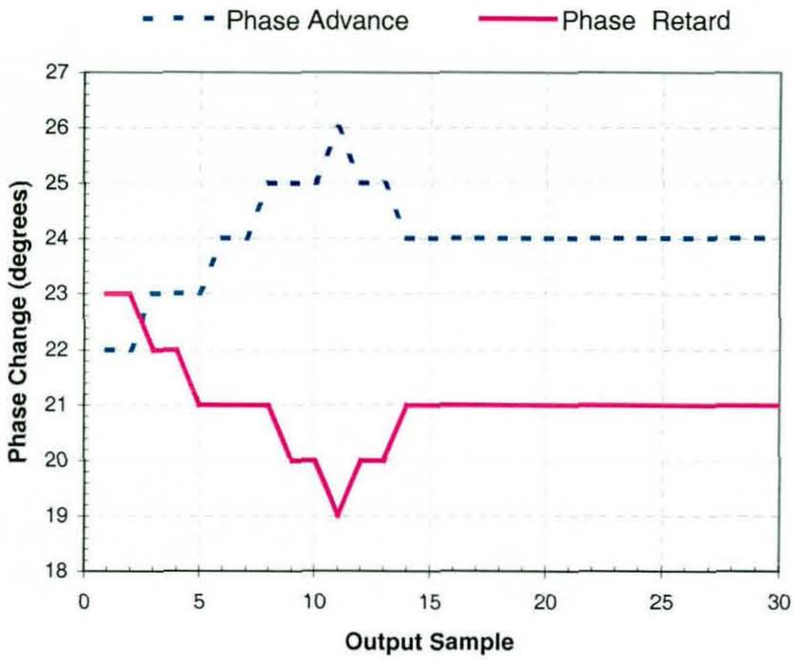


Figure 5.7 Continuous Phase Change

5.5.3 Spread Spectrum

An important observation above shows that for certain data bit patterns the transmission is a single tone. In the above situation it maybe advantageous to add a data whitening process prior to transmission that ensures spread spectrum transmission. The definition of spread-spectrum system is the following:[5.22]pp.719

- The signal occupies a bandwidth much in excess of the minimum bandwidth necessary to send the information.
- Spreading is accomplished by means of a spreading signal, often called a code signal, which is independent of the data.
- At the receiver, dispreading is accomplished by the correlation of the received spread signal with a synchronised replica of the spreading signal used to spread the information.

Standard modulation schemes such FSK and PCM also spread the spectrum of an information signal, but they do not qualify as spread-spectrum systems since they do not satisfy all of the above conditions.

Spread-spectrum systems were initially developed for military applications to prevent jamming of wireless communication links. By distributing the signal energy over many more coordinates than typically modulation schemes, system have been developed that have Low Probability of Detection (LPD). Only the intended receiver can detect the signal's existence by possessing a synchronised replica of the spreading signal. To any receiver without a replica of the spreading signal the signal appears to be buried in the noise.

Spread-spectrum signal can be used to improve the uncertainty in time-of-flight measurements as the measure of uncertainty, Δt, is proportional to the rise time of the pulse, which inversely proportional to the bandwidth of the pulse signal. The larger the bandwidth, the more precise one can measure the range. This clarifies the purpose of transmitting *gold codes* at the head of a packet, not only for range measurement, but also symbol synchronisation.

Direct-Sequence Spread-Spectrum (DS/SS) is a spreading technique whereby the carrier wave is first modulated by the data signal, then the data-modulated signal is again modulated with a high-speed (wideband) spreading signal. DS spread-spectrum technique is often used in conjunction with CDMA to achieve multi-access communications. Development of DS-CDMA in the cellular radio mobile communication industry has encouraged research into similar systems for the underwater environment.[5.14] [5.16][5.27][5.41]

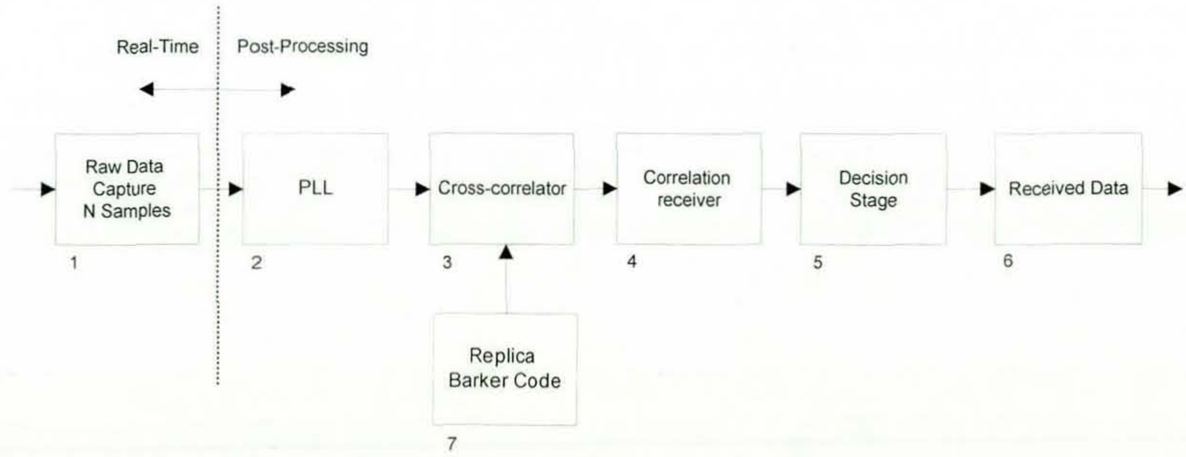


Figure 5.8 Receiver Signal Flow diagram

The receiver operates in the passband; hence the signal is not mixed down to enable less intensive baseband processing. Although baseband signal processing is less intensive the signal has to be mixed using a mixer, which requires extra components and signal processing. Figure 5.8 shows seven receiver sections that operate on the received signal.

1. Raw Data Capture - the raw passband signal is captured by an ADC and is stored to SRAM in real-time; no signal processing is performed during data capture.
2. Phase Lock Loop (PLL) – the PLL synchronises the receiver to the captured waveform, enabling coherent processing of the captured waveform.
3. Cross-Correlator -the cross-correlator is used to detect the Barker code header that provides symbol synchronisation and also an accurate time-of-arrival measurement.
4. Correlation Receiver – the correlator, which is also known as a matched filter is used to demodulate the data. In the case of binary detection, the correlation receiver can be configured as a single matched filter or product integrator, with the reference signal being the difference between the binary prototype signals. The output of the correlator is fed directly to the decision stage. For QPSK $(M = 4)$ there are four matched filters, with the output being fed into the decision stage.
5. Decision Stage – the decision stage chooses the largest correlator output, i.e. the best matched signal.
6. Received Data - this is the binary data packet received
7. Replica Barker Code – this is the chosen code that is used in the header of all communication packets.

5.5.4 Simulated BER

To test the modulation scheme and the encoding and decoding algorithms the communication scheme was simulated on a Personal Computer (PC). As explained in the System Design chapter, the embedded 186 processor is based on the same internal architecture as that of common PCs, so the actual encoding and decoding functions written for the embedded system can be run on a standard PC to reduce simulation time. Figure 5.9 shows five sets of data; *Math* is the evaluation of Equation (5.19) and trace *csb50* is a simulation of the modulation scheme described with correlated start-bit detection and data decoded at 50% thresholds, as evaluated mathematically in Equation (5.19). The *csb50* trace and *Math* trace are coincidental indicating that the simulations are represented correctly mathematically and vice versa.

Various threshold levels were simulated and it was found as expected that the 50% threshold gives the lowest BER. Obviously, due to the data packet design the start-bit is the single most important part of the data packet. If the start-bit is incorrectly detected, positional data cannot be calculated without error and encoded data cannot be decode correctly because symbol lock has not been achieved As discussed, the probability of start-bit detection can be increased by the use of *gold codes* instead of a single bit.

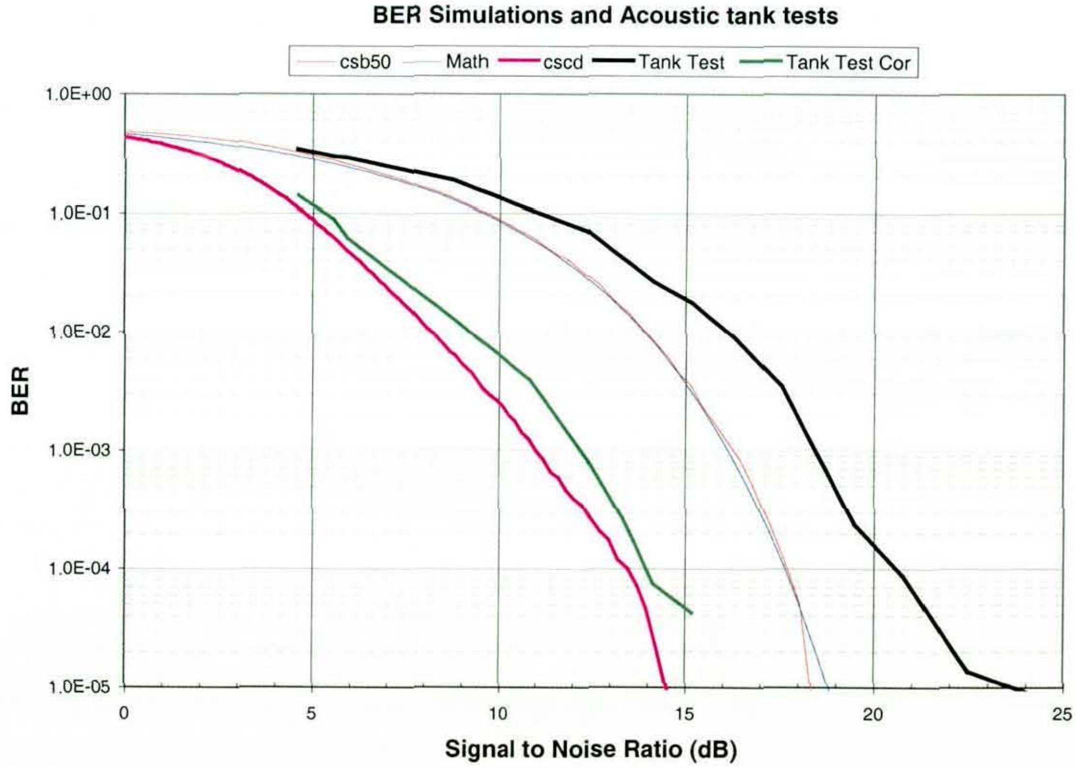


Figure 5.9 Simulated and Acoustic Bit Error Rate (BER) for different configurations

Another simulated data series displayed, 'cscd' shows the improved BER performance achieved by correlating the received data signal with a replicate of the possible phase change options, i.e. correlation receiver. The detector/demodulator uses the largest output of the correlator *z(Ts)*; in normal BPSK where there are equal-energy antipodal signals, the detector can decide on the basis of:

$S_1(t)$ if $z(Ts) > 0$

and $S_2(t)$ otherwise

The additive white gaussian noise (AWGN) to the signal in the simulations has a mean of zero ($\mu = 0$). Hence, cross-correlating a signal with noise against a replica signal without noise over x samples averages the noise, hence the effects of the noise tends to zero for large values of x.

5.5.5 Acoustic Tank Tests

The acoustic test tank at Loughborough University enables limited acoustic testing of various modulation techniques. The tank dimensions are 9m x 5m x 2m. The 2m depth limits the communication packet length due to reverberation from the bottom, surface and walls.[5.46] Sensible positioning of the hydrophones in the tank enables the transmission of a *clean* direct signal with a duration of 1ms before reverberations interfere with the received signal. The simulations and mathematical error calculation give a maximum achievable error rate performance, but they do not take in to account the effects of the hydrophones, the properties of the medium, receiver and transmitter circuitry.

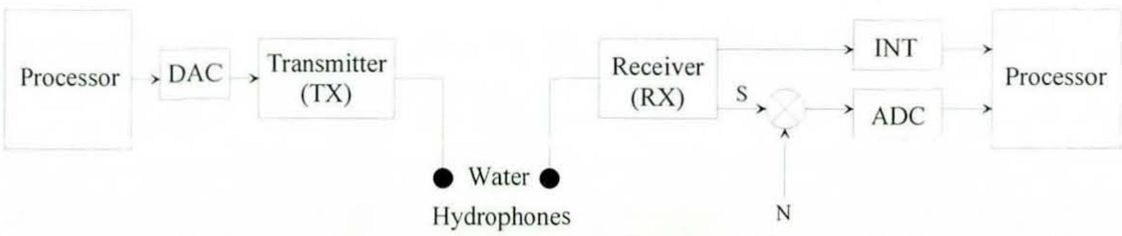


Figure 5.10 Block diagram of equipment configuration for BER tests

Short 8-bit packets of pseudo-random data were sent between two units four times a second. Figure 5.11 shows the reverberations of a signal in the test tank for a typical experimental set-up shown in Figure 5.13 . This indicates why a packet was only sent every 250ms, to ensure that the BER data was not corrupted by an increase in background noise from the previous packet transmission. For the experimental results to coincide with the mathematical and simulated results the type of noise and noise level must be known. Practically measuring the noise level of the system plus any added noise is not a simple task, as the noise level at the ADC is what is required for the BER plot. If WGN is added to the water, the noise would be band-pass filtered by the hydrophone and receiver circuitry. The additive noise is now not Gaussian and differs from the mathematical calculations. To keep the mathematics as simple as possible, WGN is injected at the ADC of the receiver, hence by-passing the filtering characteristics of the hydrophone and receiver circuit as shown in Figure 5.10

For the BER tests, two transponder units were used, a transmitter and a receiver. In the early BER tests the transmitter and receiver were also interconnected with a RS232 serial cable, which allowed the transmitter to communicate the data via the serial link prior to acoustic communications, hence the receiver just compared the serial data with the received acoustic data. There was a problem with this set-up as a bit error on the serial communication link would be flagged as an acoustic error. However, the probability of a serial link error is several orders of magnitude less than the acoustic error probability. To preclude this potential problem, later BER tests were conducted without the serial link. To perform BER tests without the serial link the receiver was synchronised to the same pseudo-random seed, hence after each transmission the next pseudo-random number is transmitted, allowing the receiver to check automatically the received data for errors. The receiver records the number of bit errors in SRAM and downloads the data to a PC via the serial cable up on request. This autonomous operation is important, as during low additive noise level tests, the test period can be several days before enough errors are detected. For a BER measurement to be considered acceptable, a minimum of ten detected bit errors is required, e.g. with a BER of 10^{-4}, 100,000 bits have to be transmitted for a single bit error, hence one million bits for the data to be considered statistically valid. Eight bits are transmitted every 250ms, so for one million bits to be communicated takes over eight and a half hours. As can be seen, the time for a single point on the BER graph takes a long period to obtain.

5.5.6 Test Procedure

Noise Measurement

The system self calibrates upon initialising, so that it can correct for any DC offset in the input circuitry. During self-calibration the input noise level at the ADC is measured by sampling the received background noise (no communication packets being received), the input buffer of sampled noise is corrected so that the mean is zero. The RMS of the buffer is then found.

$$mean = \frac{1}{N}\sum_{k=0}^{N-1} n_k \qquad (5.23)$$

Where N is the number of samples and n_k are the sample values.

The AWGN generator has a mean of zero; hence the mean value calculated in Equation (5.23) is due to a dc-offset in the receiver circuitry and is constant for the duration of the test. The noise sample n_k need to be normalised so that the mean of the samples is zero.

$$\eta_k = n_k - mean \qquad (5.24)$$

For k = 0, 1,2,..,N,

The RMS noise at the ADC was calculated by:

$$n_{RMS} = \sqrt{\frac{1}{N}\sum_{k=0}^{N-1} \eta_k} \qquad (5.25)$$

Due to the number of samples taken, this way of measuring the system noise yielded the most accurate results. Using a noise generator, WGN was added at the ADC, as shown in Figure 5.10 . Once the required level is set, the system is then activated. The receiver measures the noise level and stores the value in SRAM so that it can be downloaded later. Periodically the receiver will re-measure the noise level approximately 200ms after receiving a transmission to ensure that the transmitted signal has subsided.

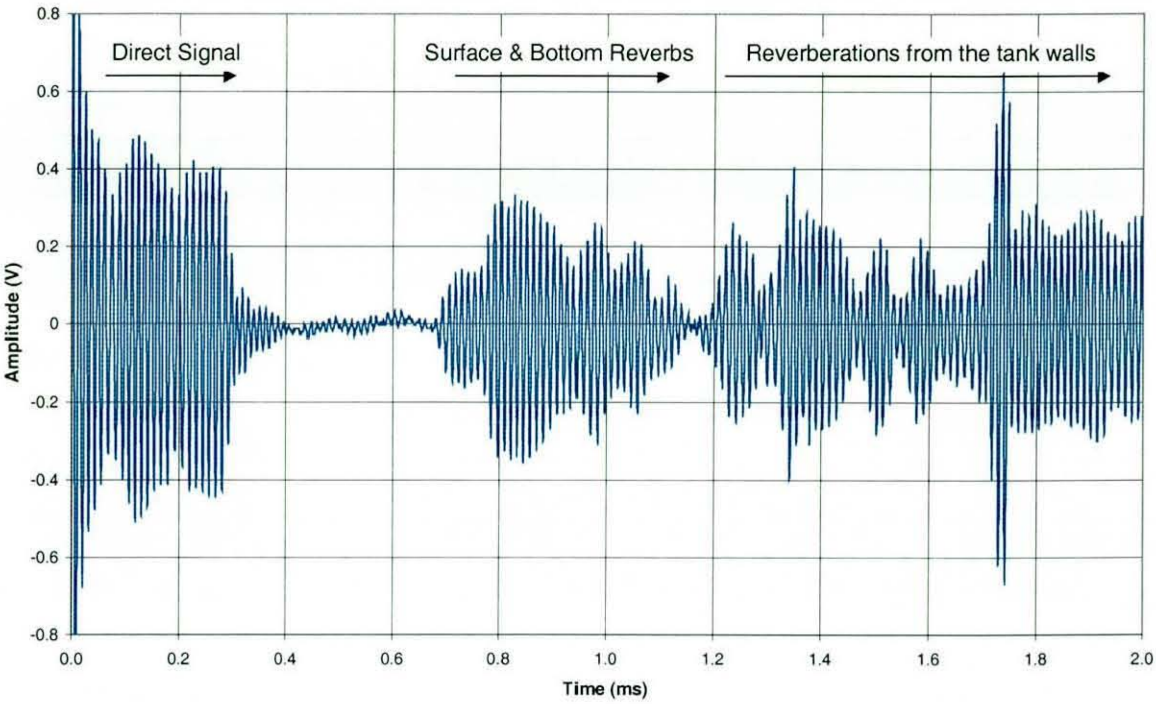


Figure 5.11 Test tank reverberations, following a data packet transmission

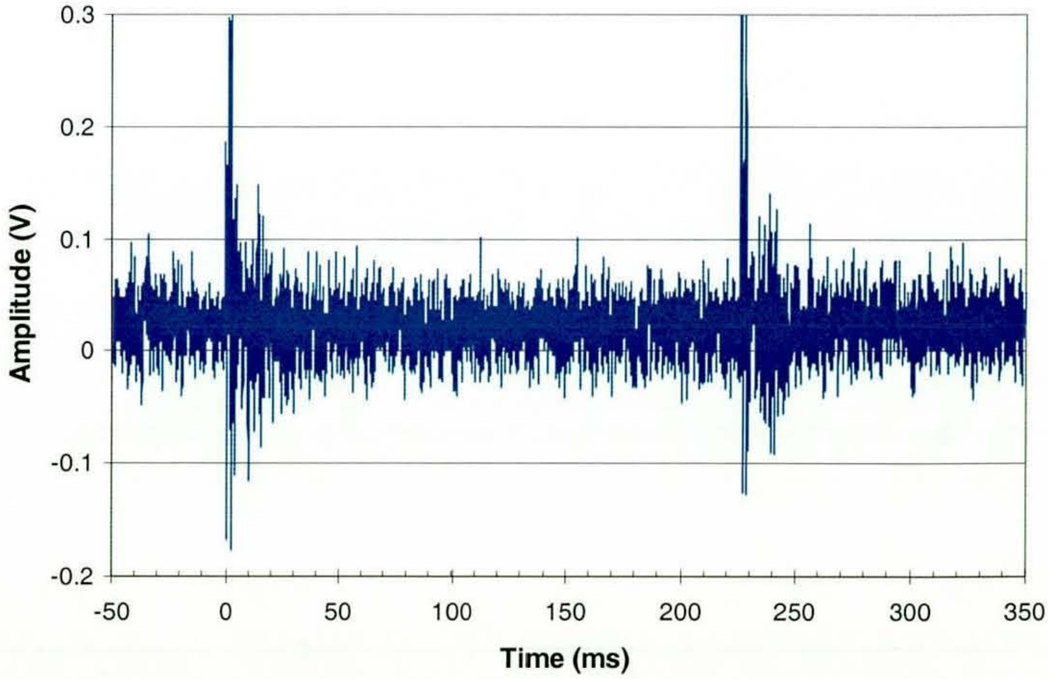


Figure 5.12 Two 8-Bit data packet transmission

Experimental Setup

The projector and receive hydrophone were positioned mid-water and spaced approximately 1m apart. Positioning the hydrophones as above allows a packet length of ≈800μs before inter-symbol interference occurs due to surface and bottom reverberations. Figure 5.13 shows the experimental setup (not to scale) and indicates the primary reverberation paths that cause ISI if the packet length is longer than 800μs.[5.46] Figure 5.11 shows a received 8-bit data packet in the acoustic test tank and the corresponding boundary reverberations. The amplitude fluctuation in the direct signal is due to the phase modulation technique used and the band limiting effects of the channel, which includes the hydrophone. Figure 5.12 shows the tank reverberations over a longer period and clearly indicates that the transmitted signal has subsided after 200ms.

The transmitter then generates pseudo-random 8-bit numbers and transmits a new number every 225ms. The receiver also generates pseudo-random numbers from the same seed, hence the transmitter and receiver are synchronised. If the receiver does not receive a packet then it will increment to the next pseudo-random number after a period of 225ms.

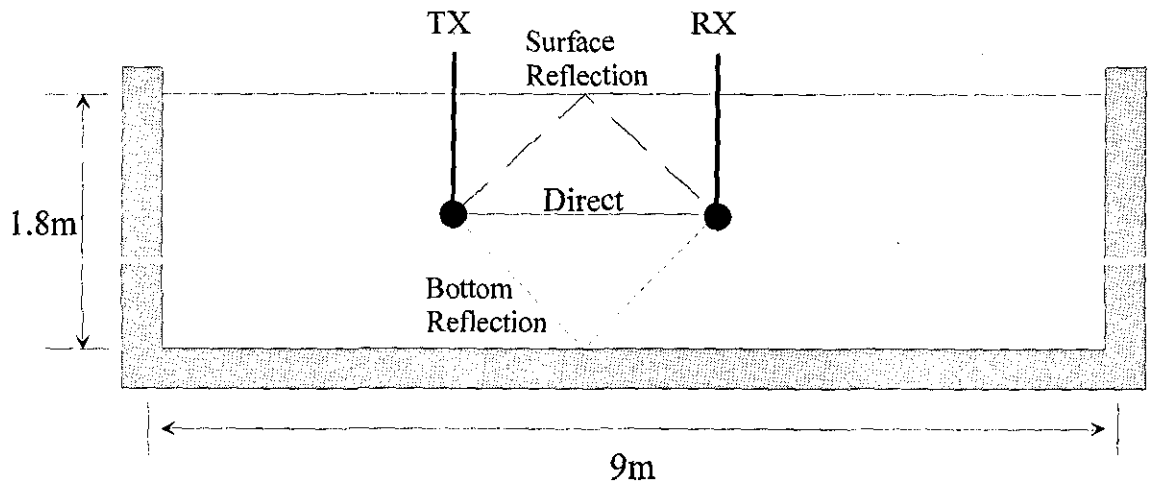


Figure 5.13 Tank test experimental setup

5.5.7 Results

The acoustic BER plot shown in Figure 5.9 indicates the number of bit errors for an increasing noise level. Considering only the Math, csb50 and the Tank test plots, it can be seen that the simulated data matches perfectly with the Mathematical calculation as expected. The tank test plot follows the same contour as the maths but has around a 2-3dB difference. As expected the practical acoustic BER test yield a slightly lower performance than the maths. Also, the plot starts to tail off, as the signal-to-noise ratio is limited by the system's self-noise.

The BER plot has two other trace: 'cscd', which means correlated start bit and correlated data bits, and tank test cor, which is the same as the previous acoustic BER tests but with every data bit correlated against a possible replica. Correlating each data bit improves the signal-to-noise performance as the correlation process averages the noise. The improvement is in the order of 3-6dB, which is to be expected with a 30-point cross-correlation per bit. [5.31]pp 196

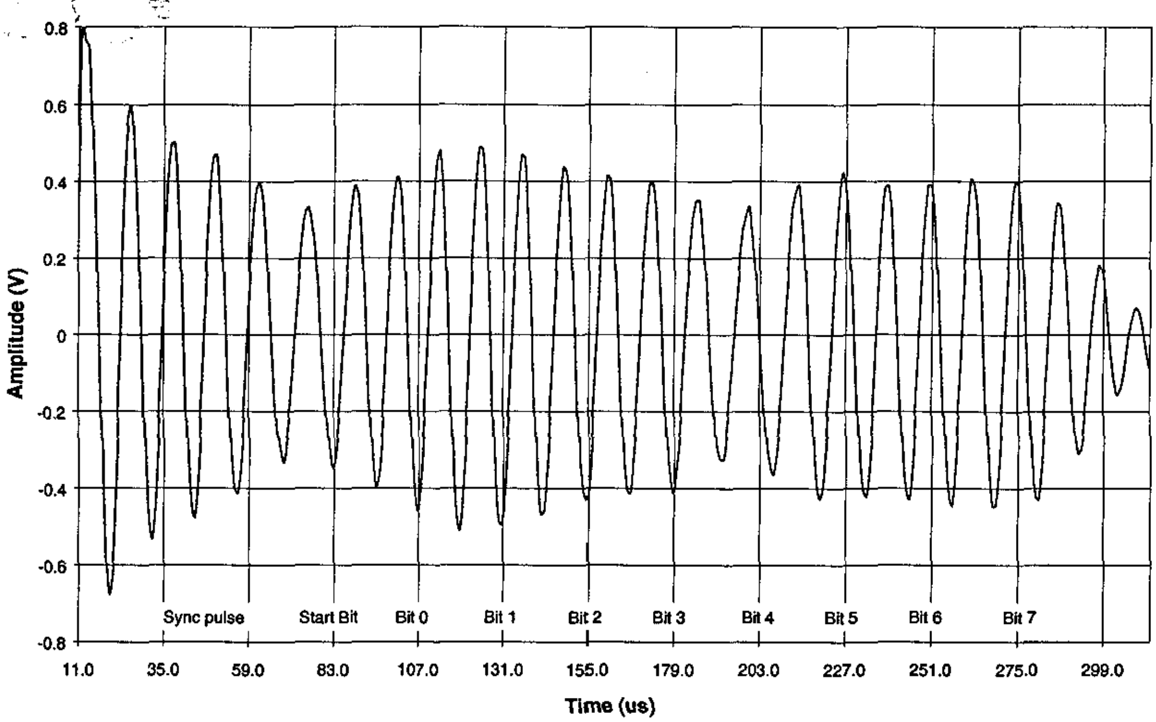


Figure 5.14 Communication test data packet

The performance of this scheme is good enough for the system to function in relatively low noise environments, which is a good feature of its autonomous operation. The performance figures from the plot show that for a BER of 10^{-3} the signal to noise ration for normal and correlated data decoding is 18dB and 12.5dB respectively.

Figure 5.14 shows an 8-bit data packet received during the tank BER tests. The large amplitude at the beginning of the data packet is due to the AGC but the data is easily decoded from the startbit. Each bit is represented by two cycles that are defined on the plot. Bits [0-3] are 1's, as there has been no change from the reference startbit. For bit4, there has be a phase change of -90deg, indicating that a logical bit change has occurred, hence bit 4 is a 0. For bit 5, there has been another phase change of -90deg, so bits 5-7 are 1's. The data Byte encoded is therefore 0xEF or 240dec. It is also clear to see the slight amplitude fluctuation of the envelope at the phase changes. There would be significantly more amplitude fluctuation of the envelope for both standard BPSK and QPSK.

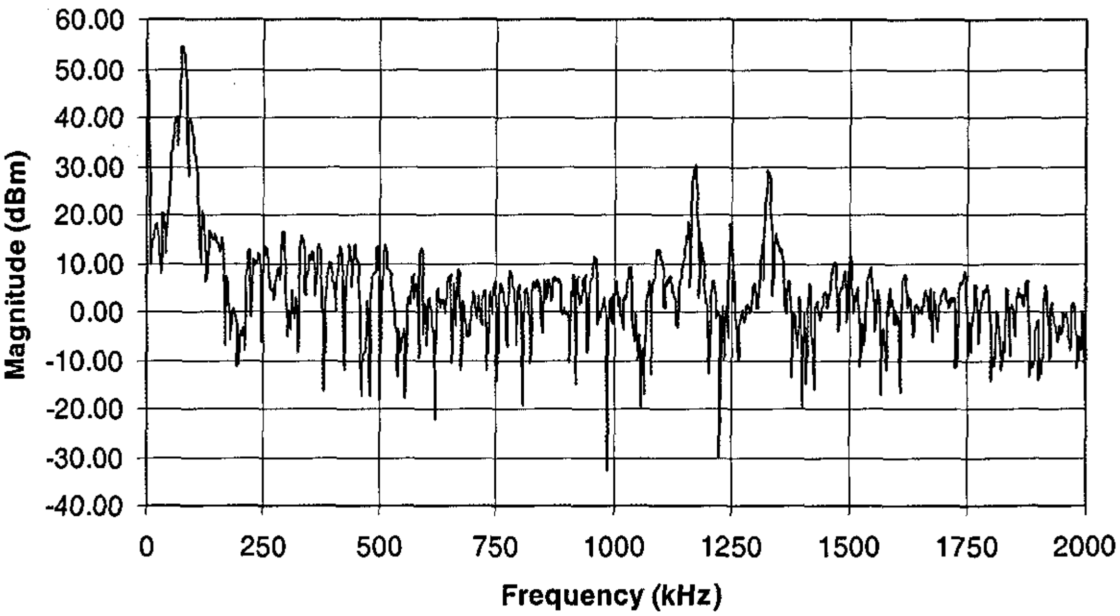


Figure 5.15 Spectrum of the transmitted waveform, include sampling frequency

The full spectrum of a typical data transmission measured at the output of the DAC is shown in Figure 5.15. The output sampling frequency f_s (f_s = 1.25MHz) is clearly shown along with the spectral peaks at $f_s \pm f_c$ (f_c = 80kHz). These spectral peak are >20dB down on the carrier frequency and due to the high output clock frequency this energy will be greatly attenuated by the transducer and the channel. A zoomed view of the frequency spectrum band of interest is shown in Figure 5.16. The bandwidth of the modulation scheme measured is approximately 12kHz. As expected the spectral tails drop away faster than a QPSK or BPSK signal, but the main lobe is wider. For this reason MSK may not be the preferred choice for narrowband links. The low spectral side lobes of MSK might be the preferred choice for multiple-carrier systems, because its low spectral side-lobes help to avoid excessive Adjacent Channel Interference (ACI).[5.22]

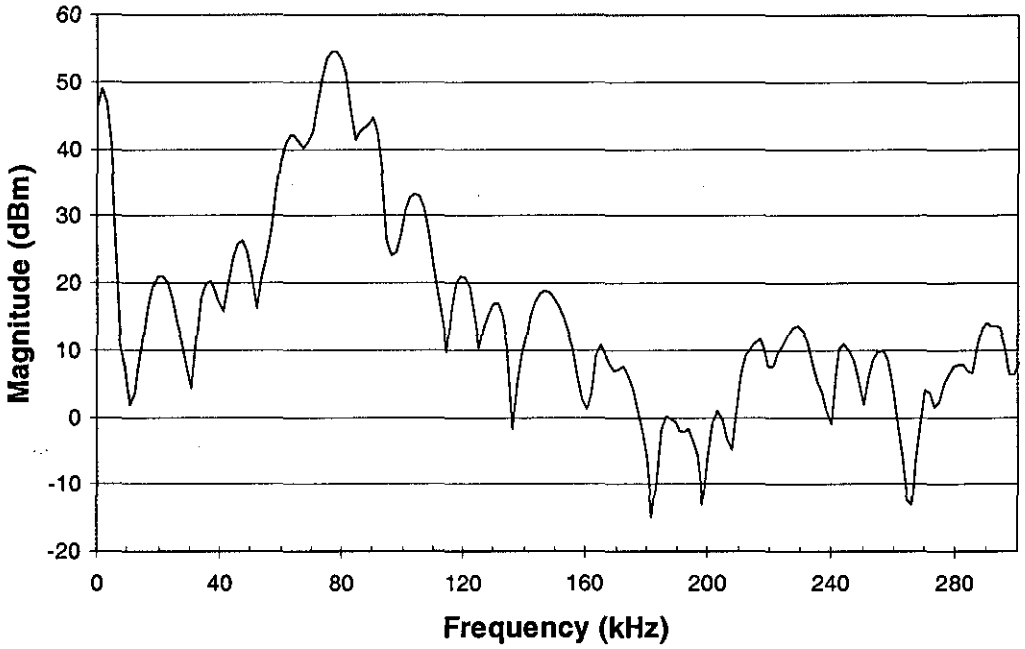


Figure 5.16 Signal spectrum

5.6 Modulation Scheme 2

This scheme is a further development of the CPOPSK described above and moves towards a CPOQPSK scheme; hence each symbol represents two data bits. To create CPOQPSK from the previous scheme requires the possibility to advance the phasor 90° as well as retard it 90°, hence the phasor in Figure 5.6 can rotate both clockwise and anti-clockwise. In the previous scheme the phase change was introduced through the whole symbol period, that is 90° over two cycles. Modulation scheme 1 achieves the same data rate as standard OQPSK, however the decoding of the data is performed twice as much, as each symbol period represent 1 bit of data rather than 2 bits.

Symbol 0	**Symbol 1**	**Data**		
Phase	*Phase*	*Bit 0*	*Bit 1*	*Bit 2*
0	0	0	0	0
0	-90	0	0	1
0	+90	0	1	0
-90	0	0	1	1
+90	0	1	0	0
-90	-90	1	0	1
+90	+90	1	1	0
-90	+90	1	1	1
+90	-90	Start	Start	Start

Table 5.6 Three bit encoding

In a conventional OQPSK scheme there are three possible phase changes, 0° or ±90° per data bit and two data bits per symbol. As QPSK indicates there are only four valid phasor positions 0°, ±90° and 180°, hence the above scheme is not efficient. By taking scheme 1 and adding the option of a 90° advance there are now three options per symbol, with still only a maximum phase change per symbol of 90°. This symbol period can now represent one and a half data bits. So by encoding the data in blocks of three bits, the data rate can be increased by 50%, to achieve a bit rate of 55Kbits/s.

Table 5.6 shows the relationship of adjacent symbols and the corresponding three bits of data that is represented by the combination of phase changes. The data packet sizes outline in the Chapter 4, are not always divisible by three, hence there can be up to two bits of data remaining. This data is encoded by appending zeros to the data packet, hence no phase change.

This is a simple further development of modulation scheme 1 and was tested in a similar, but less vigorous, manner to scheme 1 and yielded very similar BER performance.

5.7 Conclusion

The communication techniques described in the chapter combined with the packet framing and protocols discussed in the previous chapter enable several transponders to function in the acoustic test tank. With no WGN injected into the system, the communication techniques facilitated the exchanged of short high-data content packets between transponder units without errors. When the noise level in the tank was raised by the injection of noise, errors were detected by the appended CRC. The protocols dealt with the problem of error packets and the system continued to function. The effect of increasing the noise level was that it slowed the system operation down as some data packet had to be re-transmitted due to being received with errors. It was apparent that often only a single bit error occurred in the data packet, however the whole data packet is required to be retransmitted. From the data obtain during the communication tests a relatively simple error correction coding could be developed that would enable the receiving unit to correct a single error would be advantageous.

The resultant scheme enables 69-bits of data (48-bits of data excluding the ID addresses, command and error detection CRC) to be transfer in just over 1.2ms, which equate to a burst data rate of approximately 55Kbits/sec. If multiple data packets have to be transmitted due to the environment then a further 8-bits of data is encoded into the packet transmission by differing the transmit time of the second and subsequent data packet. The first data packet is not differed in time as the arrival time of this packet is used for round-trip time measurement.

The simulation of the modulation schemes using the same platform significantly reduced the evaluation time as BER simulations in the test tank are extremely time consuming. This is due the short packet transmission with a relatively small amount of data transferred and the require deadtime following the packet transmission to allow the reverberations to die away. The general rule of thumb; which require at least 10 error to be statistically valid was abided by, hence for a single point on the graph at a BER 10^{-4} requires 100,000 bits of data to be transferred for a single error. The BER test for this single point on the graph takes approximately 8 ½ hours. The simulation does not have to wait for the reverberation to die away and can produce a result is less than 10 minutes.

The simulator is very simple and only simulates the signal waveform generation, capture and decoding in an AWGN environment, it does not show the effects created by the channel and the hydrophones. For this reason the first BPSK and QPSK schemes that appear to function well when simulated failed in practice. The simulator can be advanced by instead of a simple linear interpolation between generated waveform and capture waveform the waveform can be filter to simulate the band-limiting effects of the channel etc.

The first stage of packet detection is via a simple threshold detect on the envelope of the in-band energy. This is prone to noise causing false triggering, which causes the processor to wake-up and capture the signal at the ADC. This can be improved increasing the Q of the filter, so that only energy in a very narrow frequency band (the carrier frequency) causes the processor to wake-up. Normally the in-band energy is integrated over time, which prevents in-band transient from hitting the threshold level. However, due to the packet design and relatively short synchronisation header the processor must be triggered into capture mode after only two or three synchronisation cycles.

The second stage of the detection process is performed digitally by the processor and is used to recover phase synchronisation. The synchronisation pulses are to enable the receiver to cross correlate the received signal with a replicate waveform. The cross-correlation peak indicates the best synchronisation and the output of correlator must exceed a threshold to validate a packet has been received and that carrier synchronisation has been achieved. Now the receiver has to maintain symbol synchronisation and this is achieved by detecting the start bit(s) by cross-correlating the received waveform with a replica of the start bit(s). As carrier synchronisation has already been achieved the cross-correlation of the start bit(s) is performed only every cycle (60 samples).

5.8 References

[5.1] Urick R.J
Principles of Underwater Sound
McGraw-Hill Book Company, 3rd Edition
1983, ISBN 0-932146-62-7

[5.2] Waite A.D.
Sonar for Practising Engineers
Thomson Marconi Sonar Limited, 2nd Edition
pp., 1998, ISBN 0-9528033-1-3

[5.3] Thompson D., Neasham J., Sharif B. S., Hinton O. R., Adams A. E., Tweedy A. D. and Lawlor M. A.
Performance of Coherent PSK Receivers Using Adaptive Combining, beamforming and Equalisation in 50km Underwater Acoustic Channels
Proceedings of the IEEE, ISBN: 0-7803-3519-8/96
pp 845-850, 1996

[5.4] Freitag L., Johnson M. and Frye D.
High-Rate Acoustic Communications for Ocean Observatories-Performance Testing Over a 3000m Vertical Path
Proceedings of the IEEE, ISBN: 0-7803-6551-8/00
pp 1443-1448, 2000

[5.5] Freitag L., Grund M., Singh S., Smith S., Christenson R., Marquis L. and Catipovic J.
A Bidirectional Coherent Acoustic Communication System for Underwater Vehicles
Proceedings of Oceans' '98, Nice, France
September 1998

[5.6] Jones J. C., Di Meglio A., Wang L. S., Coates R. F. W., Tedeschi A. and Stoner R.J.
The Design and Testing of a DSP, Half-Duplex, Vertical, DPSK Communication Link
MAST II Project VERTLINK, Final Report No MAS2-CT94-0079
1997

[5.7] Zheng M., Coates R. F. W. Wang L. and Stoner R.
Underwater Acoustic Communication Utilising Parametric Transduction with M-ary DPSK Modulation
Proceedings of the IEEE, ISBN: 0-780303519-8/96
pp 1-7, 1996

[5.8] Coates, R. F. W and Wang L. S.
The Bass 600 Underwater Acoustic Communication Link
Proceedings of Electronic Engineering in Oceanography, Conference Publication No. 394
pp 111-116, July 1994

[5.9] Albonico D., Fohanno F. and Labat J.
Test of a High Data Rate Acoustic Link In Shallow Water
Proceedings of the IEEE, ISBN: 0-7803-5045-6/98
pp 1028-1032, 1998

[5.10] Goalic A., Labat J., Trubuil J., Saodui, S. and Rioualen D.
Toward a Digital Acoustic Underwater Phone
Proceedings of the IEEE, ISBN: 0-7803-2056-5
pp III-489-III-494, 1994

[5.11] Falahati A., Woodward B. and Bateman S. C.
Underwater Acoustic Channel Models for 4800 b/s QPSK Signals
Proceedings of the IEEE Journal of Oceanic Engineering, Vol 16, No 1
pp 12-20, January 1991

[5.12] Kilfoyle D. B. and Baggeroer A. B.
The State of the Art in Underwater Acoustic Telemetry
Proceedings of the IEEE Journal of Oceanic Engineering, Vol 25, No 1, ISSN: 0364-9059
pp 4-27, January 2000

[5.13] Freitag L., Grund M., Singh S. and Johnson M.
Acoustic Communication in Very Shallow Water: Results from the 1999 AUV Fest
Proceedings of the IEEE, ISBN: 0-7803-6551-8/00
pp 2155-2160, 2000

[5.14] Freitag L., Stojanovic M., Singh S. and Johnson M.
Analysis of Channel Effects on Direct-Sequence and Frequency-Hopped Spread-Spectrum Acoustic Communication
Proceedings of the IEEE Journal of Oceanic Engineering, Vol 26, No 4, , ISSN: 0364-9059
pp 586-593, October 2001

[5.15] Zielinski A., Coates R., Wang L. and Saleh A.
High Rate Shallow Water Acoustic Communication
Proceedings of the IEEE, ISBN: 0-7803-1385-2/93
pp III-432-III437, 1993

[5.16] Tsimenidis C. C., Hinton O. R., Adams A. E. and Sharif B. S.
Underwater Acoustic Receiver Employing Direct-Sequence Spread Spectrum and Spatial Diversity Combining for Shallow-Water Multiaccess Networking
Proceedings of the IEEE Journal of Oceanic Engineering, Vol 26, No 4, ISSN: 0364-9059.
pp 594-603, 2001

[5.17] Yeo H. K., Sharif B. S., Adams A. E. and Hinton O. R.
Performances of Multi-element Multi-user Detection Strategies in a Shallow-Water Acoustic Network (SWAN)
Proceedings of the IEEE Journal of Oceanic Engineering, Vol 26, No 4, ISSN: 0364-9059.
pp 604-611, 2001

[5.18] Davies J. J., Pointer S. A. and Dunn S. M.
Wideband Acoustic Communications Dispelling Narrowband Myths
Proceedings of the IEEE, ISBN: 0-7803-6551-8/00
pp 377-384, 2000

[5.19] Dunn S. M., Davies J. J. and Pointer S. A.
A Real-Time High Data Rate Acoustic Communications Receiver Demonstration System
Proceedings of the IEEE, ISBN: 0-7803-6551-8/00
pp 385-390, 2000

[5.20] Van Gijzen M. B. and Van Walree P. A.
Shallow-Water Acoustic Communication with high Bit Rate BPSK Signals
Proceedings of the IEEE, ISBN: 0-7803-6551-8/00
pp 1621-1624, 2000

[5.21] LeBlanc L. R. and Beaujean P. P.J.
Spatio-Temporal Processing of Coherent Acoustic Communication Data in Shallow Water
Proceedings of the IEEE Journal of Oceanic Engineering, Vol 25, No 1, ISSN: 0364-9059
pp 40-51, January 2000

[5.22] Sklar B.
Digital Communications, Fundamentals and Applications
Prentice Hall P T R, ISBN 0-13-084788-7
pp 1-1079, 2000

[5.23] Rice S. O.
Statistical Properties of a Sine Wave Plus Random Noise
The Bell System Technical Journal, Vol XXVII, No 1
pp 109-157, January 1948

[5.24] Kimball C. V.
Intersymbol Interference in Binary Communication Systems
Cooley Electronics Laboratory, Technical Report No 195, 3674-18-T
pp 1-187, 1968

[5.25] Darby B and Woodward
Advanced Data Communications techniques for Subsea Applications
PhD Thesis, Loughborough University
2001

[5.26] Haykin S.
Communication Systems
Wiley 3rd Edition, 1994
pp. 387, Chapter on Noise, ISBN 0-471-57176-8

[5.27] LeBlanc L. R., Singer M., Beaujean P. P., Boubli C. and Alleyne J. R.
Improved Chirp FSK Modem for High Reliability Communications in Shallow Water
Proceedings of the IEEE, ISBN: 0-7803-6551-8/00
pp 601-603, 2000

[5.28] Clark G.C, Jr. and Cain J.B
Error-Correction Coding for Digital Communications
Plenum Press, 1982
ISBN 0-306-40615-2

[5.29] Press W.H., Teukolsky S.A., Vetterling W.T. and Flannery B.P.
Numerical Recipes in C
Cambridge University Press, 2nd Edition
ISBN 0-521-43108-5, 1994

[5.30] Horowitz P. and Hill W
The Art of Electronics
Cambridge University Press, 2nd Edition
ISBN 0-521-37095-7

[5.31] Ifeachor E.C and Jervis B.W.
Digital Signal Processing, Chapter 4: Correlation and Convolution
Addison-Wesley 1st Ed., 1993
ISBN 0-201-54413-X

[5.32] Freitag L., Johnson M., Grund M., Singh S. and Preisig J.
Integrated Acoustic Communications and Navigation for Multiple UUVs
Proceedings of Oceans' 2001, MTS 0-933957-28-9
pp 1-6, 2001

[5.33] Green M. D. and Rice J. A.
Channel -Tolerant FH-MFSK Acoustic Signalling for Undersea Communications and Networks

Proceedings of the IEEE Journal of Oceanic Engineering, Vol 25, No 1, ISSN: 0364-9059
pp 28-39, January 2000

[5.34] Dawoud M. M., Halawani T. U. and Abdul-Jauwad S. H.
Experimental Realization of ASK Underwater Digital Acoustic Communications System Using Error Correcting Codes
Proceedings of the International Journal of Electronics, Vol 72, No 2, ISSN: 0020-7217
pp 183-196, 1992

[5.35] Garrood D. J.
Applications of the MFSK Acoustical Communications System
Proceedings of the IEEE, ISSN 0000-0067
pp 67-71, 1981

[5.36] Gragg R. F. and Wurmser D.
Pseudo-Doppler Resonance Phenomena in Continuous Wave Scattering from Evolving Intermediate Bubble Plumes
Journal of the Acoustical Society of America, ISSN: 2317-2318
pp 473-483, 1995

[5.37] Mole L. A., Hunter J. L. and Davenport J. M
Scattering of Sound by Air Bubbles in Water
Proceedings of the Journal of the Acoustical Society of America, ISSN: 2317-2318
pp 837-841, 1970

[5.38] Sari H. and Woodward B.
Underwater Voice Communications Using a Modulated Laser Beam
Proceedings of the IEEE, ISBN: 0-7803-5045-6/98
pp 1183-1188, 1998

[5.39] Sari H.
Underwater Acoustic Voice Communication using Digital Techniques
Ph. D. Thesis, Loughborough University
1997

[5.40] Boulanger C., Loubet G. and Lequepeys J. R.
Spreading Sequences for Underwater Multiple-Access Communications
Proceedings of the IEEE, ISBN: 0-7803-5045-6/98
pp 1038-1042, 1998

[5.41] Tsimenidis C. C., Hinton O. R., Sharif B. S. and Adams A. E.
Spread-Spectrum Based Adaptive Array Receiver Algorithms for the Shallow-Water Acoustic Channel
Proceedings of the IEEE, ISBN: 0-7803-6551-8/00
pp 1233-1237, 2000

[5.42] Stojanovic M. and Freitag L.
Multiuser Undersea Acoustic Communications in the presence of Multipath Propagation
Proceedings of Oceans', MTS 0-933957-28-9
pp 2165-2169, 1999

[5.43] Beaujean P. P. J. and LeBlanc L. R.
Spatio-Temporal Processing of Coherent Acoustic Communications Data in Shallow Water
Proceedings of the IEEE, ISBN: 0-7803-6551-8/00
pp 1625-1631, 2000

[5.44] Gomes J., Barroso V., Ayela G. and Coince P.

An Overview of the ASIMOV Acoustic Communication System
Proceedings of the IEEE, ISBN: 0-7803-6551-8/00
pp 1633-1637, 2000

[5.45] Sharif B. S., Neasham J., Hinton O.R. and Adams A. E.
A Computationally Efficient Doppler Compensation System for Underwater Acoustic Communications
Proceedings of the IEEE Journal of Oceanic Engineering, Vol 25, No 1, ISSN: 0364-9059
pp 52-61, January 2000

[5.46] Zheng Z., Woodward B. and Newborough D.
Calculating the allowable co-ordinates of acoustic transducers to avoid multipath interference in a tank
ACUSTICA - Acta Acustica, 2002

CHAPTER SIX

POSITION FIXING

6. POSITION FIXING

6.1 Introduction

In traditional nautical terms, navigation essentially implies that a ship has moved from one point to another, whereas in the underwater context, it is often expanded to include tracking, e.g. a submersible (ROV or AUV) tracked by a surface vessel throughout its deployed operation; positioning, e.g. a diving bell positioned relative to a work site for welding or repairs; and measurement, e.g. the precise surveying of the seabed or an archaeological site and distance measurement between pipe ends or points of particular interest.

To have a system that combines all of the above capabilities, and hence providing accurate distance measurement; positioning and tracking of ROVs/AUVs and divers, would be advantageous. Various acoustic and non-acoustic underwater positioning techniques have been developed to achieve these capabilities with different degrees of accuracy.[6.1]

Dead Reckoning navigation systems:

- Distance line
- Trailing wheel
- Current meter
- Doppler log
- Inertial navigation

Underwater acoustic navigation systems:

- Ultra-Short Baseline (USBL)
- Short BaseLine (SBL)
- Long BaseLine (LBL)
- Sonar
- Hyperbolic

The greatest use of SONAR (SOund NAvigation and Ranging) is in commercial echo sounders, used for depth measurements or to detect fish in the water column. The echo-sounder can be used as a navigational aid because marine charts have water depths indicated on them. Sonar systems are also used for other navigational aids, such as speed over the seabed, velocity logs using Doppler systems or imaging the seabed using scanning sonar.

Traditional methods of finding underwater objects often entail divers being sent down to search for them. In the past this has occasionally involved a diver being dragged behind a boat and relying on his eyesight to detect the object. Once the location of the object had been established a marker buoy is dropped and if possible land sightings are taken to enable the position to be easily relocated.

Advances in sonar technology and improved transducer and array design have enabled systems to pinpoint and identify objects from a surface vessel. Sonar systems exist that can track an object, such as a diver.[6.2] Old systems use a mechanically steered sonar transducer, which is physically moved to give bearing and range to an object by measuring the flight time of a transmitted pulse and its returning echo. These systems often use Time Varying Gain (TVG) amplifier to normalise the amplitude of the returning echoes with time. The sonar system produces a map of the ensonified seabed, showing the contours and objects that reflect the sound back towards the sonar receiver. Using transponders enables bright illumination to appear on the map as the transponder transmits upon detecting a sonar ping, producing a strong return signal. This tracking system is similar to the USBL or SBL systems discussed in Chapter 2: Acoustic Positioning Systems.

6.1.1 Active Positioning Systems

Active positioning systems generate an acoustic signal of a known form and at a known time. This enables the system to measure the time taken for the acoustic signal to propagate from one transponder to another. Thus, knowing the propagation time and the sound velocity in water, a distance can be calculated. Spherical methods to calculate position are commonly used in most LBL positioning systems. The position of a transponder is found by calculating the intersection of two or more spheres with two-dimension spherical algorithms. Three-dimensional spherical algorithms require three or more transponders to solve a mobile transponder's position $P(x,y,z)$. Using transponders and direct ranging, it is possible to achieve excellent position accuracy.

Two-dimensional positioning assumes that the Mobile transponder M being positioned is in the same plane as the transponder array. To position a transponder in two-dimensions requires a three-receiver array to calculate its position without ambiguities. If a single transponder transmits a ping to another transponder, which waits a short period T_d before transmitting a reply, a distance between the transponders can be calculated by Equation (6.1).

$$d = c.(\Delta t - T_d)/2 \quad (6.1)$$

where c is the velocity of sound (m/s) and T_d is the dead time between receiving a signal and transmitting a reply. The transponder can be anywhere on a circle of radius *d* from the source (assuming no transducer directionality) as shown in Figure 6.1 .

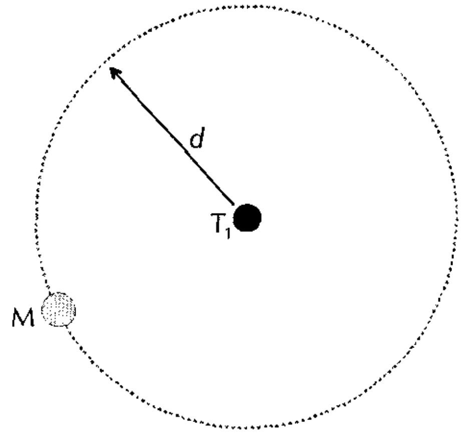


Figure 6.1 Single range measurement to a mobile (M) transponder

A positioning system that operates in this manner is a spherical positioning system and is based on the mathematics of intersecting spheres. In two dimensions the spheres can be reduced to circles if all the transponders are in the same plane. To unambiguously position a transponder in two-dimensions there needs to be a minimum of three receivers in the position-fixing array and the intersection of all three circles is the position of the mobile transponder. Figure 6.2 shows the three range circles around the corresponding transponders and the position of the Mobile unit at the point where all the circles intersect. Also indicated on the diagram are the three other possible positions if only two receivers are used (M_{12}',M_{13}' & M_{23}').

The equations for the circle around each transponder shown in Figure 6.2 are:

$$d_1 = ct_1 = \sqrt{(x+s)^2 + y^2} \quad (6.2)$$

$$d_2 = ct_2 = \sqrt{(x-s)^2 + y^2} \quad (6.3)$$

$$d_3 = ct_3 = \sqrt{(x-x_3)^2 + (y-y_3)^2} \quad (6.4)$$

The x-axis co-ordinate of *M* is found by subtracting equation (6.3) from equation (6.2) to give:

$$x = \frac{(d_1^{\ 2} - d_2^{\ 2})}{4s} \quad (6.5)$$

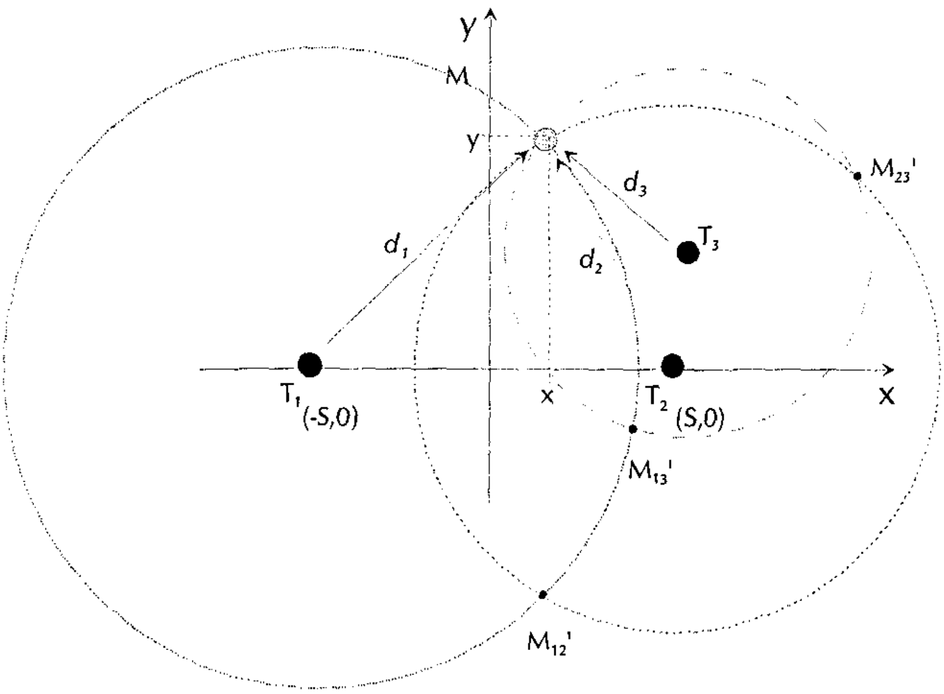


Figure 6.2 Two-dimensional absolute positioning

The y-axis co-ordinate of M is found by substituting x to give:

$$y = \pm\sqrt{d_1^{\ 2} - (x+s)^2} \quad (6.6)$$

$$y = \pm\sqrt{d_2^{\ 2} - (x-s)^2} \quad (6.7)$$

As can be see from Figure 6.2 there are two possible positions of *M*, either *M* or M_{12}, when positioning *M* with only two transponders T_1 and T_2. The third transponder can confirm whether the y co-ordinate is positive or negative by substituting y into equation (6.4).

The drawback of the solution above is that the transponders are considered to be on the x-axis and symmetrical about the y-axis, which may not be the case. However, when positioning only relative to the transponder array, the Master transponder assumes the position *(0,0)* and transponder T_2 will be positioned on the x-axis *(x, 0)*. To ensure that this is not a constraint of the system the transponder must be able to be given any x, y co-ordinates.

So, assuming transponder T_1 is at co-ordinates (x_1,y_1) and transponder T_2 is at co-ordinates (x_2,y_2) then equations (6.2) and (6.3) are re-written as:

$$d_1^2 = (x - x_1)^2 + (y - y_1)^2 \quad \text{.......} \quad (6.8)$$

$$d_2^2 = (x - x_2)^2 + (y - y_2)^2 \quad \text{.......} \quad (6.9)$$

Subtracting equation (6.8) from equation (6.9) gives the solution for x where:

$$x = \frac{\left[x_2^2 - x_1^2 + y_2^2 - y_1^2 - d_2^2 - d_1^2\right] - 2y(y_2 - y_1)}{2(x_2 - x_1)} \quad \text{.......} \quad (6.10)$$

Once x is known y can be found by using either equation (6.8) or (6.9), which is of the form:

$$ay^2 + by + c = 0 \quad \text{.......} \quad (6.11)$$

which give the solution:

$$y = \frac{\left[-b \pm \sqrt{(b^2 - 4ac)}\right]}{2a} \quad \text{.......} \quad (6.12)$$

Two-dimensional positioning algorithms are required during the baseline transponder calibration, as the position of the third transponder is calculated based on the range between the Master and first transponder detected. The first three transponders can only be positioned using two-dimensional positioning, as there are insufficient transponders to calculate a three-dimensional fix.

Once the seabed array is calibrated, the relative positions of at least three transponders are known. In the previous cases only two-transponder navigation has been used, hence there are two possible solutions. To provide a unique solution a third range is required. This range, d_3 is indicated in Figure 6.2 and provides an unambiguous solution for the position of the mobile transponder *M*.

Using spherical trigonometry as before, a unique solution is found in the form:

$$(X_2 - X_1)x + (Y_2 - Y_1)y = (P_2^2 - P_1^2)/2 \qquad (6.13)$$

$$(X_3 - X_2)x + (Y_3 - Y_2)y = (P_3^2 - P_2^2)/2 \qquad (6.14)$$

$$(X_1 - X_3)x + (Y_1 - Y_3)y = (P_1^2 - P_3^2)/2 \qquad (6.15)$$

Solving any two of these three simultaneous equations gives the desired co-ordinates for *x* and *y*. The derivation of these equations is fully documented in reference[6.1].

Often the transponders are not all at the same depth so the horizontal projection is not d_1. If d_1 is the measured distance using acoustics then the horizontal projection for an offset depth z_1 is:

$$R = \sqrt{d_1^2 - z_1^2} \qquad (6.16)$$

If the depth of the mobile device is known then the position of the mobile unit can be calculated using the two-dimensional equation above. If the mobile unit is at an unknown depth then the equations above are expanded to include the z-axis dimension:

$$(x - X_n)^2 + (y - Y_n)^2 + (z - Z_n) = R_n^2 \qquad (6.17)$$

for n = 1, 2, 3, 4...n

To obtain a solution for x,y and z from equation (6.17) it is necessary to eliminate the x^2, y^2 and z^2 terms by subtracting pairs of equations to give the form:

$$A_n x + B_n y + C_n z = D_n \qquad (6.18)$$

for n = 1, 2, 3...n.

This is a set of over-determined equations, hence four equations in three unknowns, x, y and z. A matrix solution for four equations of this type is simply:

$$Mx = N \qquad (6.19)$$

where matrix M is the observation matrix:

$$M = \begin{bmatrix} A_1 & B_1 & C_1 \\ A_2 & B_2 & C_2 \\ A_3 & B_3 & C_3 \\ A_4 & B_4 & C_4 \end{bmatrix} \qquad (6.20)$$

The constant matrix and solution matrix are:

$$N = \begin{bmatrix} D_1 \\ D_2 \\ D_3 \\ D_4 \end{bmatrix} \qquad (6.21)$$

$$x = \begin{bmatrix} x \\ y \\ z \end{bmatrix} \qquad (6.22)$$

To solve the simultaneous equations for x, y and z requires the inverse matrix M to be found and it is only possible to find the inverse of a square matrix, i.e. same number of columns and rows. To find the solution requires matrix M to be made square by multiplying both sides of equation (6.19) by M^T (M^T is the transpose of M). Thus, the following equation gives a solution for x, y and z.

$$x = (M^T M)^{-1}(M^T N) \quad (6.23)$$

The C code used for inverting a 3x3 matrix is shown in Appendix C: Postioning Algorithms. Other positioning algorithms are discussed later in this chapter that do not require matrix inversion, however trigonometry math functions (sine, cosine, cot, tan) are required.

6.1.2 Passive Positioning Systems

A passive positioning system does not transmit a signal; hence the time of flight cannot be directly measured as with an active positioning system. A passive tracking system consists of a beacon attached to a mobile unit, which pings at regular intervals, and a receiver array of known hydrophone separation. The time that the beacon transmits is unknown, so instead of measuring the time-of-flight (time between transmitting and receiving a response from a transponder) the system measures the difference in signal arrival times on each hydrophone. Spherical positioning algorithms are no longer applicable. Instead a hyperbola is formed around each pair of receivers, and the intersection of these hyperbolic loci indicates the possible solutions.

The positioning system describe throughout this thesis has been designed as an active positioning system. Most commercial underwater acoustic positioning systems operate in this mode, where direct ranges are measured and the position calculated from three or more range timings. Passive positioning is discussed here to present a solution to the expensive and time-consuming geodetic calibration procedure. This is where the positions of the transponder units on the seabed are related to one another and often to surface positioning systems such as GPS. Performing this calibration procedure enables future survey data to be precisely correlated with previous surveys even after redeployment of seabed transponder arrays, as the absolute positions are known. Algorithms for position fixing transponders on the seabed from surface slant range measurements are discussed in reference[6.5]. The problem with using slant ranges to position the seabed transponder rather than direct baseline ranging is that the sound velocity profile changes with depth; more so than with horizontal range. To achieve accurate transponder positions requires knowledge of the sound velocity profile. With direct seabed ranging, an average of the sound velocity profile is usually adequate, as the range measurements are relative to the seabed array and the sound velocity profile tends not to vary significantly over horizontal ranges[12]. The autonomous design of UAPS ensures that minimum ranges are measured; hence the diver or ROV will often be close to the seabed. The design of the system does not require range measurements to be made or relayed to the surface-positioning vessel where the largest sound velocity fluctuations in the water column frequently occur.

The concept of inexpensive long-life acoustic marker beacons is presented to enable position data from several surveys to be correlated without the requirement of a surface calibration. The acoustic beacons are deployed along with the transponder array during the first survey of the site. The transponder array passively positions the beacons in two-dimension as the transponders and beacon should be in a similar plane. The coordinates of the transponders are relative, as are all the position measured/calculated to the permanent fixed seabed beacons. The expensive transponder technology is recovered from the survey site, leaving the acoustic beacons in place. The design of the acoustic beacon described in chapter 3, is extremely efficient and can operate for over two years on a single *D*-cell battery depending on the transmission repetition rate. The transmission repetition rate for the baseline marking can be relatively slow; for example, once every 60 seconds. However, the inter-ping duration is dithered to ensure that two or more beacons do not remain synchronised, as this would prevent the transponders from accurately positioning them. By using passive positioning techniques, knowledge of the exact time the beacon transmitted is not required, as the beacon is positioned using the arrival time difference between the seabed transponder array. Figure 6.3 shows the time-of-arrival differences, t_{12}, t_{13} and t_{23} and the corresponding hyperbolic loci. The intersection of these loci will be the source of the transmission, i.e. the beacon. All baseline marker beacons are positioned independently. From multiple position observations a least squares fit of the beacon's position can be calculated to increase the accuracy and confidence in the data. This enables accurate post-processing of the data as the position of the baseline is accurately known.

[12] The sound velocity can vary over horizontal ranges due to ocean currents, sediment transfer etc, however generally the sound velocity varies with depth substantially more than with horizontal range.

When revisiting a survey area where acoustic marker beacons have been deployed, the transponder array can be randomly re-deployed in the survey area. The transponder array will position-fix the beacons relative the redeployed transponder array. The position datum can then be re-aligned to the previous surveys by transposing the position coordinates of the transponder array, as shown in a later section: Position Conditioning. The positions calculated by the seabed transponder array can now be directly compared with previous surveys, as the positions have been calculated with the same datum.

Two-dimensions

For two-dimensional positioning using passive techniques:

$$d_t = \Delta t_{12} \times c$$

from the geometry $d_t = d_1 - d_2$

Using Pythagoras' theorem:

$$d_1^{\,2} = (b+x)^2 + y^2 = (d_t + d_2)^2 \quad (6.24)$$

where b is half the baseline separation of the two receivers,

$$d_2^{\,2} = (x-b)^2 + y^2 \quad (6.25)$$

Eliminating the unknown distances d_1 and d_2 gives:

$$(b+x)^2 + y^2 = \left\{d_t + \left[(x-b)^2 + y^2\right]^{1/2}\right\}^2 \quad (6.26)$$

$$2bx - d_t^{\,2} = 2d_t + \left((x-b)^2 + y^2\right)^{1/2} \quad (6.27)$$

Squaring equation (6.27) and re-arranging it gives the form:

$$1 = \frac{x^2}{j^2} + \frac{y^2}{k^2} \quad (6.28)$$

where

$$j^2 = \frac{d_t^{\ 2}}{4} \text{ and } k^2 = \frac{4b^2 - d_t^{\ 2}}{4}$$

Equation (6.28) is the definition of a hyperbola, consisting of two curved surfaces symmetric to each other as shown in Figure 6.3 . Due to the squaring of equation (6.27), the resultant expression does not contain the necessary information to determine which surface describes the locus. This ambiguity can only be removed by repeating the positioning of the source using another pair of receivers and finding the intersection of the two loci.

The position of the source can be found by calculating the loci for various receiver pairs; the intersection of these pairs produces the solution. These multiple equations can be solved in a number of ways, e.g. by absolute algorithms, symbolically, tabular, etc.[6.10]

The equations for calculating the co-ordinates of any point P(x,y,z) with respect to an arbitrary datum point are shown in reference [6.8]. This set of equations can enable the position of a baseline marker beacon to be position in three-dimensions. To position an acoustic beacon in three-dimensions requires four seabed transponders. However, the baseline marker beacons will be at a fixed location on the seabed, hence a known relative depth. This simplifies the algorithms presented in [6.8] and a full derivation is shown in reference[6.9]. Making the substitution $ct_1 - ct_2 = c\Delta t_{12}$ from equations (6.2), (6.3) and (6.4) yields:

$$c\Delta t_{12} = \sqrt{(x-x_1)^2 + y^2} - \sqrt{(x-x_2)^2 + y^2} \quad \text{.......................................} (6.29)$$

$$c\Delta t_{32} = \sqrt{(x-x_3)^2 + (y-y_3)^2} - \sqrt{(x-x_2)^2 + y^2} \quad \text{...............................} (6.30)$$

From these two equations the position P(x,y) can be found as shown in the derivation in reference[6.9]. There are a few exceptions to the theory were the algorithms fail to compute, these are:

- $\Delta t_{12} = \Delta t_{13} = 0$

- $\Delta t_{31} = 0, \Delta t_{12} \neq 0$
- $\Delta t_{12} = 0, \Delta t_{31} \neq 0$
- $\Delta t_{31} = -\Delta t_{12}$, i.e. $\Delta t_{32} = 0$

For a seabed array shown in Figure 6.3 the baseline ranges d_{12}, d_{13} and d_{23} can be measured using direct ranging and the relative positions of the three transponders can be calculated. The baseline acoustic beacon transmits a contention-based emission, which ideally would at a different frequency to normal operation. This would enable the transponder to only listen to beacon emissions if baseline re-alignment is require and once re-aligned the transmission can be mask out.

Normally in a passive positioning system the receivers are all connected electrically together enabling the navigation processor to directly compute the time-of-arrival difference between receivers. In the system describe the Slave transponder are required to relay the arrival time back to the Master transponder to enable the position of the beacon to be calculated. The extended command set allows the Master to instruct the Slave transponders to relay the arrival time of the beacon *ping* after a set deadtime, which can be individually set by the Master to ensure that the various Slave replies do not coincide (see Appendix B). The procedure of relaying the beacon acoustic ping to the Master is very similar to the NDM state discovery response (see chapter 4).

Once the master transponder has receiver all of the relayed arrival times the time difference t_{12}, t_{13} and t_{23} can be calculated using Equation (6.31):

$$t_{12} = (t_{2R} - t_{relay} - t_{Range12}) - t_1 \qquad (6.31)$$

Where t_{12} is the difference in the arrival time of the beacon ping at transponder 1 and 2. The arrival time of the relayed transponder 2 ping is t_{2R} and the deadtime between receiving the original beacon ping and transmitting a relay response is t_{Relay}. Also, the relay response will incur an acoustic propagation $t_{Range12}$, which is known as the range to all transponder is found during the array calibration procedure. Figure 6.3 shows the acoustic propagation of a ping (dashed lines) and the arrival time of the wave front at each transponder.

In a typical array deployment the distance between transponders will be between 5 and 50 metre. The range is advantageous as the time-of-arrival difference between transponders is likely to be large relative to the clock drift and timing response errors, which are quantified later in this chapter.

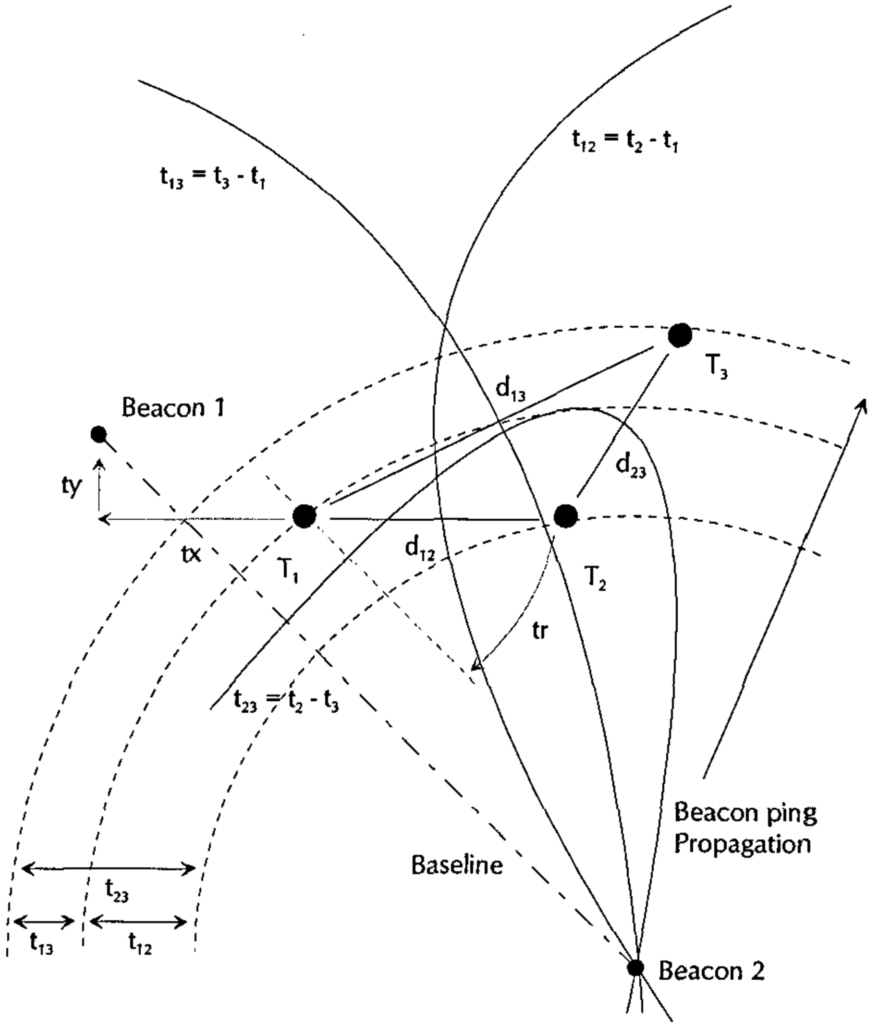


Figure 6.3 Passive positioning of acoustic beacons

Implementing the algorithm on the embedded system is easy due to standard math libraries and floating-point emulation. Depending on the mathematical precision, that is, 16-bit, 32-bit or 64-bit the time to compute the formulae outline in reference [6.9] at maximum precision (double) on the embedded 186 processor is less than one second. The accuracy of the two-dimension positioning algorithms is presented in reference [6.9] and shows a maximum error of 34mm and a minimum error of 2mm. These are attributed to the rounding and truncation errors inherent in any computer program. With double precision calculations these errors can be minimised, insomuch as the timing errors predominate.

6.2 System Accuracy

When reviewing underwater position-fixing systems, care must be taken to ensure that the operational accuracy is never compared, inadvertently or otherwise, with the ideal performance envisaged by the manufacturers. It is therefore essential that the two main types of errors - systematic and random - are extracted from the technical documentation.

Systematic errors are those that are inherent in the design of the system and can usually be attributed to known causes such as the velocity of sound, bending or refraction of the acoustic paths. Random errors are unpredictable and can be due to changes in the environment or the equipment, such as fluctuation in the ambient noise due to a passing boat or movement due to tidal flow. It is difficult to make allowances for random errors, as they can only be described statistically. When reviewing the accuracy of an underwater positioning system the repeatability of a measurement is often the most important characteristic, hence the ability to return to a selected point with a high degree of confidence. Thus, an underwater positioning system that has large systematic errors, provided they are constant and small random errors, should provide good repeatable position data.

Position errors are introduced in several forms:

- Environmental errors, i.e. variations in the sound velocity due to temperature salinity and pressure.
- Non-linear acoustic ray paths.
- Image interference due to reflection.
- Multiplicative errors for speed of sound in slant-range measurements.
- Earth curvature errors when operating long baseline systems (not an issue in medium baseline system as developed here).
- Baseline errors in initial calibration and survey.

- Square-law errors from quadratic equations.
- Transducer motion during interrogation.
- Electronic errors, i.e. timing errors: start-bit detection, response times and processor clock drift.

A system should be designed to eliminate or significantly reduce errors introduced by the last four points. The other errors can then be reduced by adequate monitoring of the water column and careful deployment of the system.

The system is capable of timing the arrival of a signal to the resolution of the processor's timer clocks, which is 100ns. However, all times are referenced to the start bit(s), hence the maximum resolution is the sampling rate, 200ns (5MHz). To ensure that the system performance is not degraded significantly by software timing response errors and clock drift all five modules were calibrated. The following tests were performed to determine the system errors.

6.2.1 Clock drift test

The amount that a 40MHz crystal clock drifts varies from unit to unit, and the maximum drift per second will give the minimal attainable accuracy due to clock drift. The 40MHz clock used on the processor module is a small ceramic resonator, with a specified tolerance of 15ppm (parts per million). To evaluate the clock drift between units, the processors were all programmed with identical firmware code. Hence all units were identical, in both hardware and software. The units were evaluated in pairs by connecting an interrupt line from both of the units to a push button switch that when pressed interrupted both units simultaneously. Upon an interrupt, the time jiffies (ms) that the processor had been running (32 bit, hence wraps after approximately 50 days) is displayed along with the timer 2 count (100ns) value that is used to update the jiffies. The units were left to run for a considerable amount of time and it was found that the maximum clock drift error between the five units was 1.7μs/second.

During normal system operation it is rare that the response time between receiving a valid packet and transmitting a reply will be greater than 1 second; the exception being during the discovery period where there are 16 time-slot windows. Depending on the response time-slot, inter-packet period and response time, it could be as long as 15 seconds. However, only distance estimation is made during the discovery period, which is then used during the baseline calibration procedure to determining whether a unit has remained stationary during the discovery and ID allocation period. When positioning a mobile unit the maximum period in which clock drift can be a factor is generally less than 500ms. This is the dead-time following a command packet capture and equates to a maximum timing error of 0.85μs. Hence, the clock drift of the system in the worst conditions corresponds to less than 1.5mm distance error over a 500ms period.

6.2.2 Cable Calibration

As discussed in Chapter 4, Network and Protocols, the software is complex and ensuring that the software transmits precisely (to within 1 timer tick, 100ns) is difficult to measure even with a high-tech digital storage oscilloscope. The software timing is further complicated by the requirement to transmit at ¼ the normal clock-speed, which is achieved by dividing the main processor clock by 4. The divider on the embedded 186 processor divides all the timer increments by 4. If this is not adjusted following a transmit routine, the transmitting processor will *lose* time.

To evaluate both the timing correction and the precision that packets are sent and received, two units were connected together electrically, instead of acoustically. This removes the acoustic propagation delay (assuming the EM propagation delay is negligible over the interconnecting cables (<4ns)) the system should calculate a propagation time of zero. The cables connected the output of the DAC buffer (the power amplifier was not used during these tests) to the injected input (the injected input by-passes the pre-amplifier front end as it would be overloaded, although it is protected by diode clamps). As with a received acoustic signal the analogue module generates an interrupt and the signal is captured via the 12-bit ADC.

Performing the cable tests enabled the precision of the system to be evaluated in various operational states. As new software was added to the system, for example, multiple packet software layer, the cable test ensured that the software introduced no timing errors.

6.2.3 Acoustic Calibration

As discussed earlier in this chapter the repeatability of position fixing is probably the most important factor in a positioning system. Also, when considering a positioning system with sub-centimetre accuracy, which uses one-inch ball hydrophones, there is a need to know where on the hydrophone to measure. The cable calibration test does not include all hardware section, so errors in the system may not be detected. The acoustic calibration test the whole system in a known configuration.

A metal hydrophone holder was machined precisely with two slots (to within 0.001mm) 1 metre between hole-centres, and this was suspended in the acoustic test tank. The system then performed several measurements of the distance between the two hydrophones using the acoustic propagation time. Figure 6.4 shows the hydrophone arrangement.

The standard deviation of the repeated measurements was taken to indicate the repeatability; cable tests yielded a standard deviation of 57.9ns and the acoustic tests 98.5ns. These values indicate high repeatability and are consistent with the earlier clock drift experiments (response time 300ms, hence possible clock drift error of 500ns). The maximum deviation from the mean during the acoustic and cable tests was 800ns.

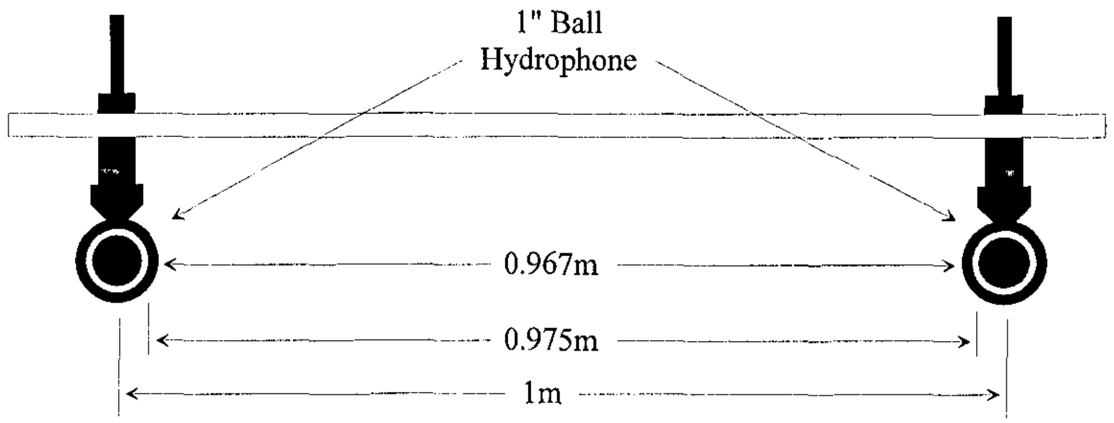


Figure 6.4 One-metre acoustic calibration test

To calculate distance from a known one-way propagation time, the sound velocity of the water must be known. The speed of sound in water depends upon the temperature, pressure (depth) and salinity.

A variety of empirical formulae exist for its calculation, and the following formula was used (Leroy)[6.3]:

$$c = 1492.9 + 3(T-10) - 6x10^{-3}(T - 10)^2 - 4x10^{-2}(T - 18)^2$$

$$+1.2(S - 35) - 10^{-2}(T - 18)(S - 35) + H/61 \quad (6.32)$$

Where c = speed of sound in m/s

T = temperature °C

S = Salinity in parts per thousand (ppt)

H = depth in metres

The temperature of the water in the acoustic test tank was 15.9°C, measured using a digital thermometer. The water in the tank is fresh, hence a salinity of zero and the tests were conducted at a depth of one metre. Inputting the values into the empirical equation (6.32) yields a sound velocity of 1460.88 m/s.

The mean one-way propagation time during the test was 0.678684 ms, which equates to a distance of 0.991476 metres, using the sound velocity calculated. This indicates that the sound wave does not propagate from the centre of the ball nor the outer periphery, but 5mm from the centre. The sound is considered to radiate from the periphery of the peizo-electric sphere, with the following factors accounting for the 16mm error:

- Acoustic test tank salinity not zero
- Thermometer error (a calibrated thermometer was not used)
- Errors introduced by the empirical formulae
- Mechanical positioning (although the mounting frame was machined to a high tolerance, the hydrophones could still be misaligned.)
- Peizo-electric element misaligned during manufacture.

The accuracy of the system has been evaluated by the three tests: clock drift, cable calibration and acoustic calibration. From the measurements made, adjustments or *fiddle factors* can be incorporated into the software to compensate for the errors. However, this was not necessary as these tests show that the systematic timing errors are less than 10^{-6} seconds.

The next area where errors are introduced is by finite variable length in the math used to calculating the position. These errors are evaluated both practically and through simulations of the positioning algorithms.

6.2.4 Calibration Algorithm

The position of a mobile unit is calculated with respect to the seabed-based transponder network. Therefore, the system calculates the distances between each transponder by measuring the round trip time of a *timing response packet* and hence, with a known sound velocity, the distance can be calculated. Consider the simplest network of transponders to give three-dimensional position in Figure 6.5 , Transponder T_1 being the Master assumes co-ordinates (0,0,0). The second transponder T_2 is positioned a distance d_{12} along the x-axis, hence $(d_{12},0,0)$. The final transponder in this case, T_3, could be in either position T_3 or T_3' the mirror image. The position of T_3 can be found by first calculating the angle ϕ using the cosine rule or by using equations (6.5) and (6.6).

$$\phi = \cos^{-1}\left(\frac{d_{12}^{\ 2} + d_{13}^{\ 2} - d_{23}^{\ 2}}{2 \times d_{12} \times d_{13}}\right) \qquad (6.33)$$

Once ϕ is known the co-ordinate position (x,y) of T_3 is calculated:

$$x = d_{13} \cos \phi \qquad (6.34)$$

$$y = d_{13} \sin \phi \qquad (6.35)$$

The disadvantage of using the cosine rule is that the cosine and sine math libraries have to be included in the software and are of considerable size. Also, cosine and sine functions take significant processor time; similarly a square root is computationally intense. The time to calculate a cosine or sine can be significantly reduce by using a ROM look-up table, however at the expense of memory.

The two possible positions of T_3 are either in the positive y-axis or negative y-axis co-ordinate set. If all positional data is calculated to the same datum, i.e. the seabed array, it does not matter whether transponder T_3 is considered to be in the positive or negative y-axis co-ordinate set. The position of the mobile transponders will be calculated to the same co-ordinate set, hence the relative position of a mobile transponder to another will be correct.

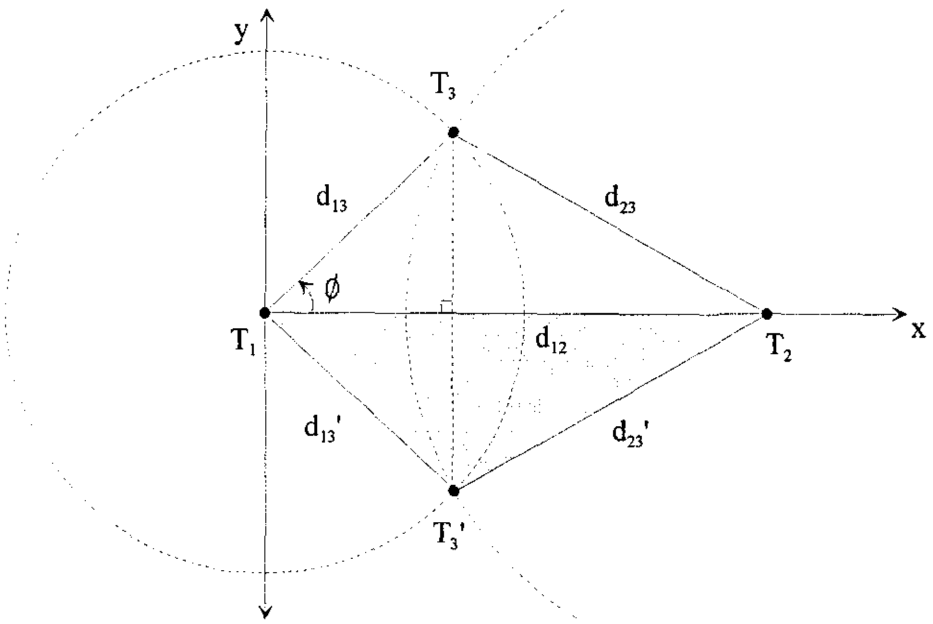


Figure 6.5 Transponder calibration configuration

The x,y co-ordinates enables multiple units to be referenced to the same grid, which has no orientation to North or absolute position as in surface positioning systems such as GPS. However, a surface vessel could be manoeuvred above the seabed transponder array to allow the seabed position data to be correlated and aligned with GPS position. The surface vessel traversing above the seabed transponders perform this alignment procedure, during which time the transponders are tracking, calculating and transferring the positional data to the surface vessel. Correlating the acoustic and GPS position information during this procedure enables the seabed position data to be translated in the longitude, latitude and altitude or depth data.

6.3 Controlled Acoustic Tests

To assess the performance and accuracy of the system designed and developed throughout out this thesis it is pointless deploying the transponders at random positions in a body of water and moving a mobile unit around. This only shows that the system is calculating a position, which may have no relevance as to its true position relative to the array. An open water trial is advantageous to show the functionality of the system once the system accuracy or repeatability has been proven in controlled conditions. The acoustic test tank and Loughborough University was convenient although due to its dimensions not ideal for positioning and communication testing. Two moveable gantries spanning the width of the tank provide secure hydrophone mounting points and enable the hydrophones to be moved linearly along the length of the tank. Also, each gantry has an integral moveable platform, enabling hydrophones to be moved in a linear manner perpendicular to the main gantry's movement. The acoustic tests utilised this parallel movement capability to prove the millimetric accuracy obtainable with the system.

6.3.1 Two-Dimensional Positioning

The calibration of the seabed transponder array is a two-dimensional problem when there are only three transponders. Three transponders can be considered to be in the same plane, even though this plane may not be parallel to the surface. To evaluate the positioning accuracy of the system, two-dimensional positioning was performed in the acoustic test tank. Three-dimensional positioning is an extension of the two-dimensional position algorithms, which can be simulated. The master transponder T_1 always initially considers itself to be at co-ordinates *(0,0)* and transponder T_2 at $(d_{12}, 0)$, where d_{12} is the distance between the Master transponder and T_2, as shown in Figure 6.6 The position of transponder T_3 is then positioned in the *x,y* co-ordinate set by measuring the time-of-flight between T_1 & T_3 and T_2 & T_3 hence distances d_{13} and d_{23}. To test the position accuracy and more importantly the repeatability, accurate repeatable positions are required. To test this, two of the transponders that form the baseline were attached at fixed location to ridged poles mid-water. The third transponder was fixed to a ridged mount on the moveable platform that bridges the tank.

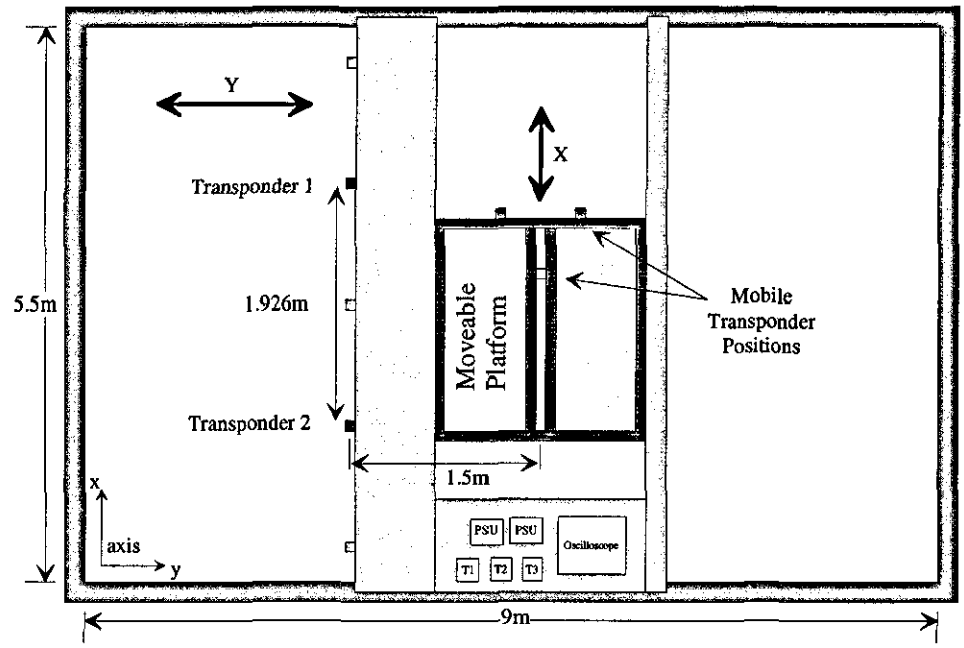


Figure 6.6 Acoustic tank test configuration

The arrangement of the transponders was so the platform moved parallel to the base array, i.e. T_1 & T_2, as shown in Figure 6.6 . The system was configured to continually perform the baseline array calibration procedure, i.e. positioning of transponder T_3 relative to the Master, T_1 and transponder T_2.

The platform was then slowly moved from one side of the tank to the other while the system calculated the x,y position of T_3. This test was repeated three times to observe the repeatability of the position measurements. Only single measurements were made at each position, hence no averaging or least squares solution derived from multiple observations. Figure 6.7 shows the tank test position measurements for the three runs and the baseline transponder positions. Figure 6.8 shows a zoomed view of the position measurements; a trend line has been added to indicate that transponder T_3 was moved at a slight angle and not parallel to the baseline transponders. To qualify this test set-up error; a 10mm y-axis deviation occurred for a 1.5m x-axis movement, i.e. an angle of 0.38°. From the zoomed plot in Figure 6.8 the correlation of the three runs is excellent giving a repeatable position error of less than 1mm.

The sub-millimetre repeatable position accuracy of the calibration procedure shows the timing precision of the system. A reliable, repeatable micro-second timing accuracy using multi-packet command and response transmissions highlights the timing precision achieved by correlating onto the start-bit (time datum) rather than relying on integrating the pulse energy and threshold level-detect circuit to generating an interrupt that can experience variable interrupt latency and noise jitter.

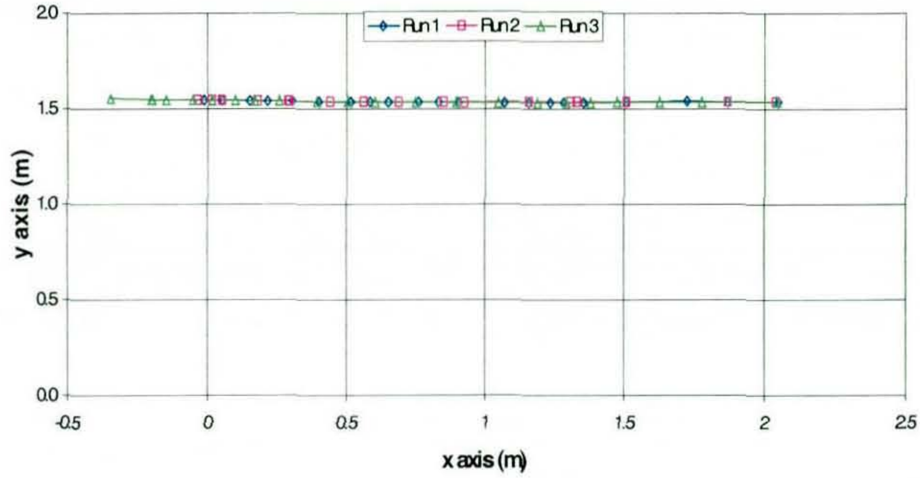


Figure 6.7 Positioning accuracy during tank tests

The test results presented are for ideal conditions, that is, excellent signal to noise ratio, constant sound velocity profile etc.

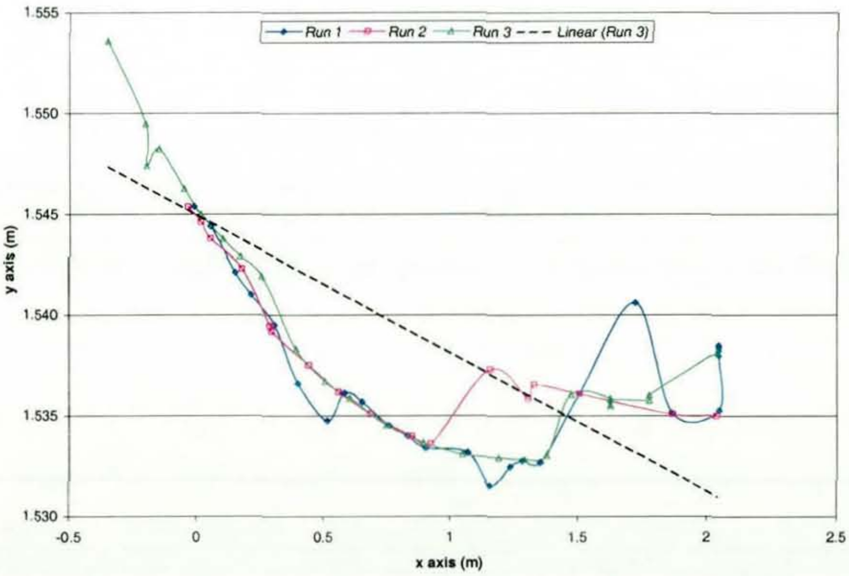


Figure 6.8 Zoomed view of the position accuracy tests shown in Figure 6.7

The next two-dimensional test positioned the mobile transponder at different y-axis locations on the moveable platform, as indicated in Figure 6.6 . The two baseline transponders are shown on the plot; these positions were the same for all the two-dimensional positioning tests performed in the tank.

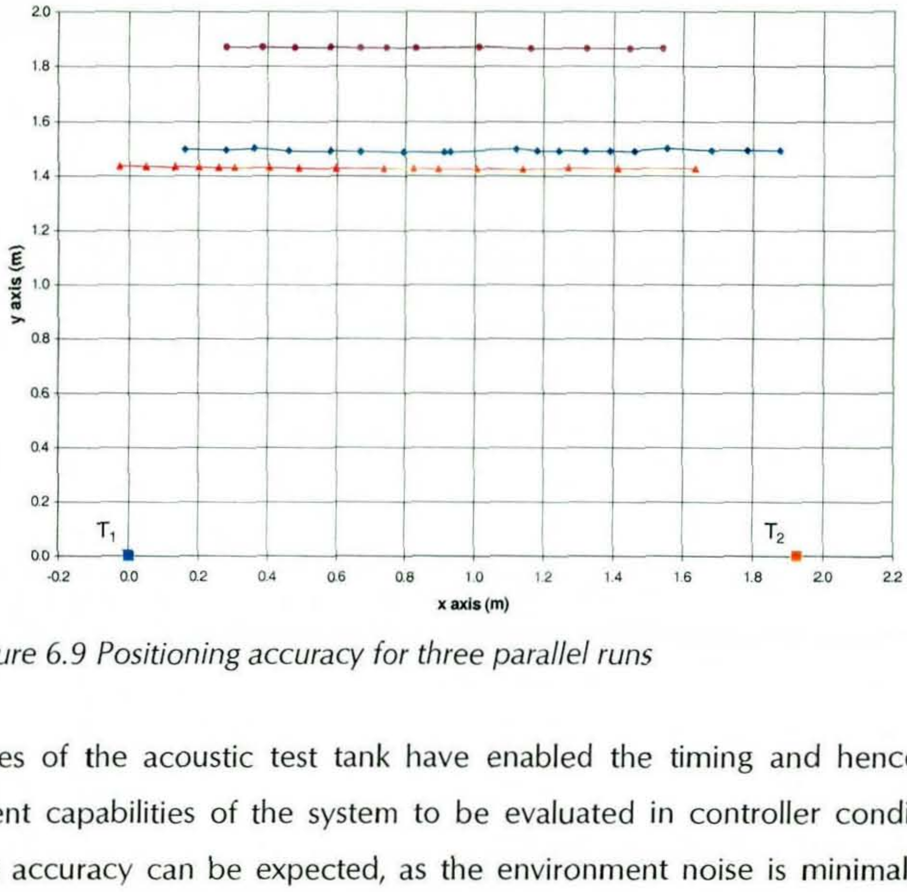


Figure 6.9 Positioning accuracy for three parallel runs

The confines of the acoustic test tank have enabled the timing and hence distance measurement capabilities of the system to be evaluated in controller conditions. The remarkable accuracy can be expected, as the environment noise is minimal, however due to the multipath environment the system has to transfer data using multiple packet transmissions.

The controlled tests have shown that the various components of the system are functioning correctly. The hardware is capable of communicating and precisely measuring the transmit and receive times of acoustic packets. This also confirms the software network access protocols, packet encoding/decoding, framing and error detection layer are all functioning correctly.

6.4 Simulated Data

The geometry of the of the acoustic test tank, especially the depth is a serious limitation when attempting to assess the performance of a positioning system. However, from the data obtain by performing the previous controlled tests it is possible to simulated the three-dimension positioning algorithms accurately for the system. Using MatLab it is possible to simulate the performance of the algorithms, for various positions.

Firstly, the simulation calculates the slant ranges to the seabed transponders *(T_1, T_2, T_3)* for a known position *P(x,y,z)*. The slant ranges from position *P* to the seabed transponders are calculated as shown in Figure 6.2 and Equation (6.36).

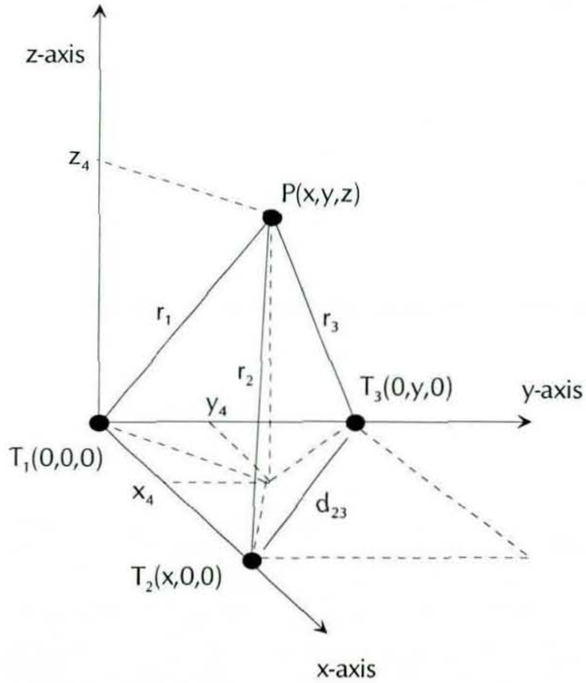
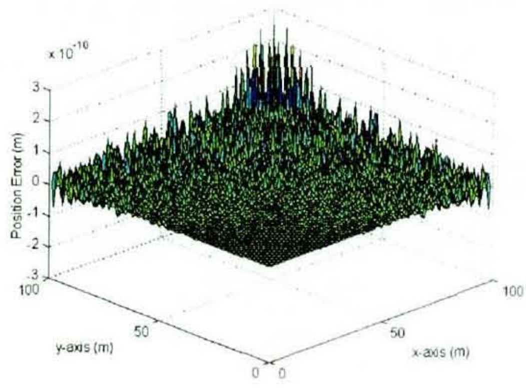


Figure 6.10 Slant range calculations

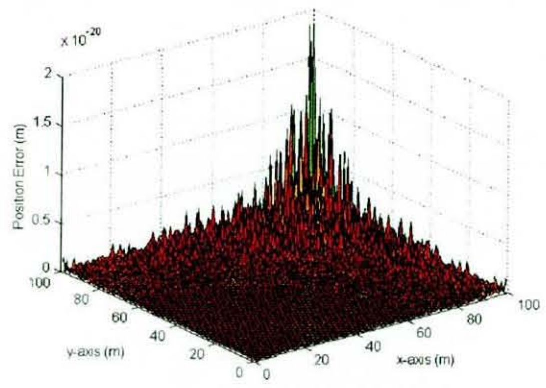
$$r_1 = \sqrt{x_4 + y_4 + z_4} \qquad (6.36)$$

Where r_1 is the slant range from Position *P(x,y,z)* to transponder *T_1(x,y,z)*. The distances between the seabed transponders are also required and are calculated in a similar manner. The positioning algorithms are used to re-compute the actual position *P*; the calculated position and actual position are compared and an error vector is calculated. The error vector magnitude is plotted on the *(x, y)* coordinate set for a particular depth z.

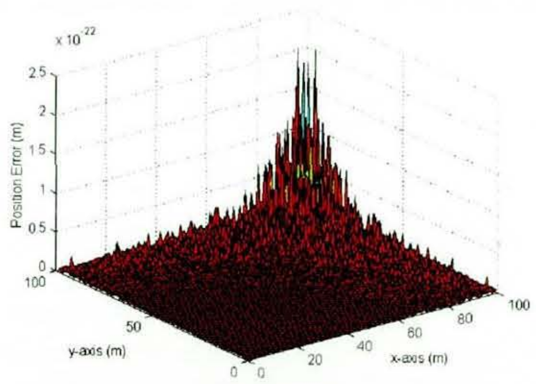
To evaluate the algorithms the precision of calculations were not reduced to 32 or 16bit resolution, as the AMD 186 processor is capable of performing double precision (64bit) mathematical operations if required. The following surface error plots shows a quadrant of a 200x200-metre area. Each quadrant is not identical due to the seabed array asymmetry. The seabed transponder positions used during the simulations are: $T_1(0,0,0)$, $T_2(5,0,0)$, $T_3(0, 5, 0)$ in a right angle triangle as shown in Figure 6.10 .



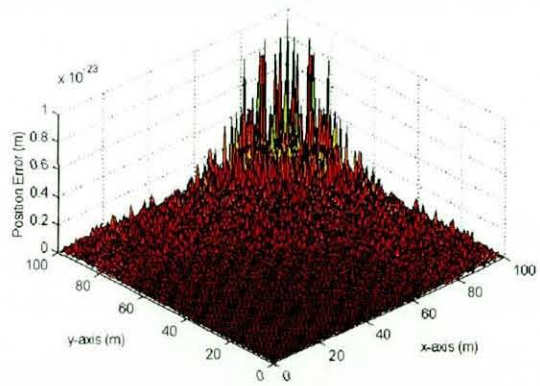
Depth z = 0



Depth z = 1m



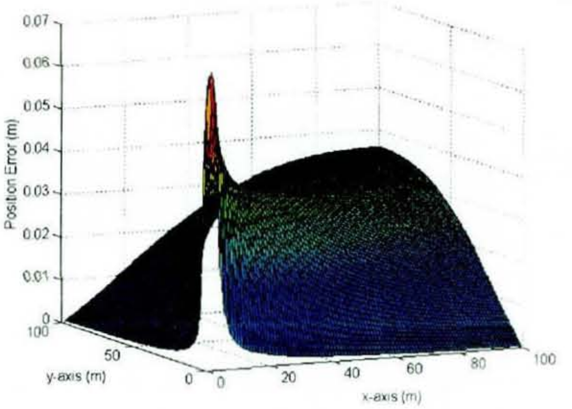
Depth z = 10m



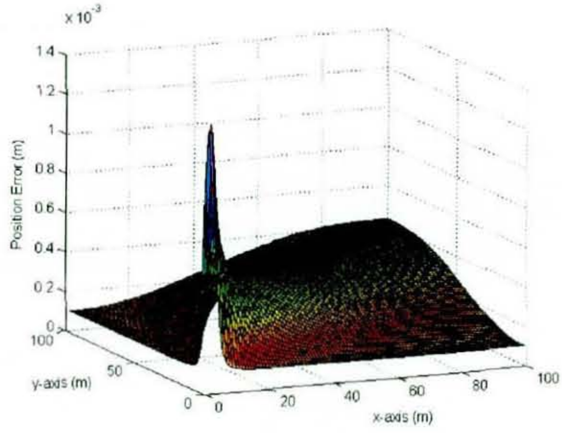
Depth z = 50m

Figure 6.11 Positioning error vector magnitude (high-resolution)

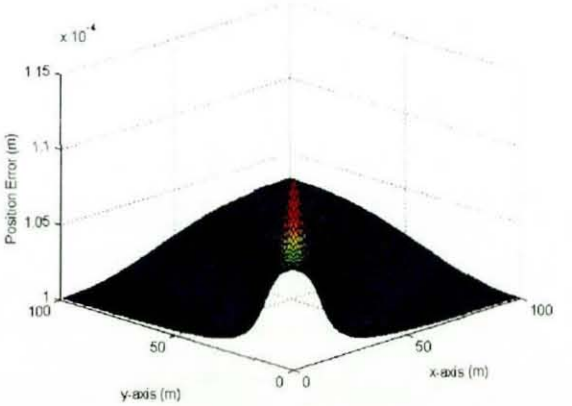
The position errors shown are extremely small (10^{-22}), which is due to the high-resolution that MatLab can calculate cosines, sine and square roots. Although the errors will scale if the resolution is reduced, hence the position errors when close to the plane of seabed transponders will be several orders of magnitude greater than when a metre or so away. On the embedded system performing high-resolution calculations are time consuming. Reducing the calculation resolution will increase the magnitude of the error vector. The compound errors introduced by insufficient resolution mathematical operations have to be assessed in the context of the system. The system can measure millimetric ranges (10^{-3}); hence the error introduced by the positioning algorithms should not degrade the systems performance further. Calculating the position using 16-bit and 32-bit math enables sub-millimetric precision to be calculated on the embedded system in the prescribed time. (See Appendix C, Processor Profiling).



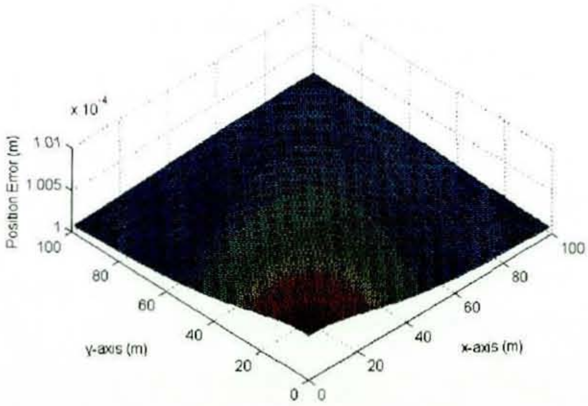
Depth z = 0



Depth z = 1



Depth z = 10



Depth z = 50

Figure 6.12 Positioning error vector magnitude with 0.01m slant range errors

Introducing small errors into the slant ranges measurements increases the position error significantly and highlights the mathematical precision required. The plots shown in Figure 6.12 indicate the error introduced by a 1cm slant range measurement error. As can be see the position errors due to the resolution of the calculation is in significant.

6.4.1 Three Dimensional Positioning

Once the positions of the three deployed transponders are established the position of a diver in three-dimensional space can be calculated, assuming the diver cannot go below the transponders. This ambiguity, if it occurs, can be removed if a surface vessel is present as this then acts as a fourth transponder, with a known z-position. Figure 6.13 shows a typical location of a diver above three randomly deployed transponders.

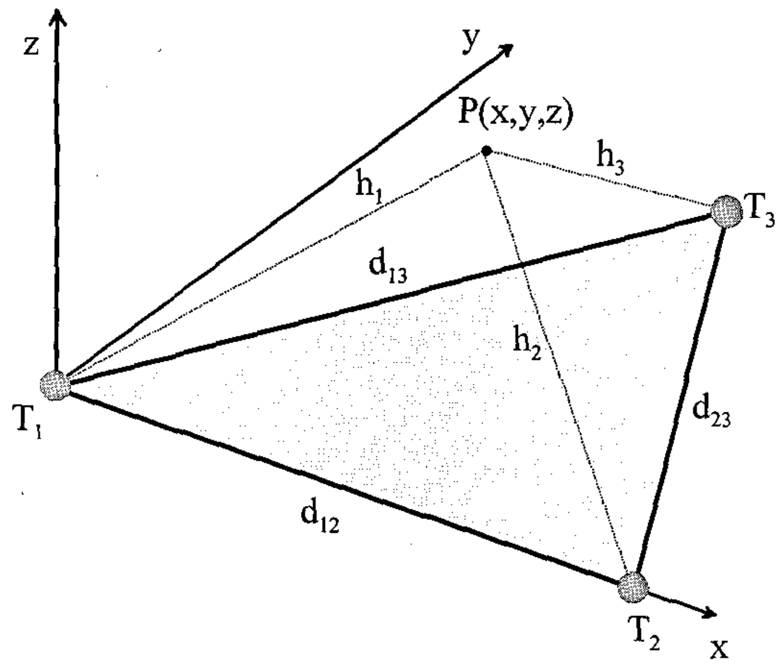


Figure 6.13 Three-dimensional position

All of the distances d_{12}, d_{13}, d_{23}, h_1, h_2 and h_3 can be calculated by measuring the time of flight of the two-way path between the transponders and the diver. As all the distances are known the problem can be visualised in two-dimensions, as in Figure 6.14 and described in detail elsewhere [6.4].

The position $P(x,y,z)$ obtained is relative to the plane of the transponders and not relative to a horizontal plane, unless by chance the seabed is exactly level. Each transponder is fitted with a pressure sensor to measure the depth, which allows the angle of the plane relative to the horizontal plane to be calculated. If the transponder plane is then rotated into the normal plane, the diver's position can also be rotated by the same amount. The result is that the diver's position is relative to the surface. This make more sense as the diver's depth profile will be relative to the surface and not to his position in the x,y plane.

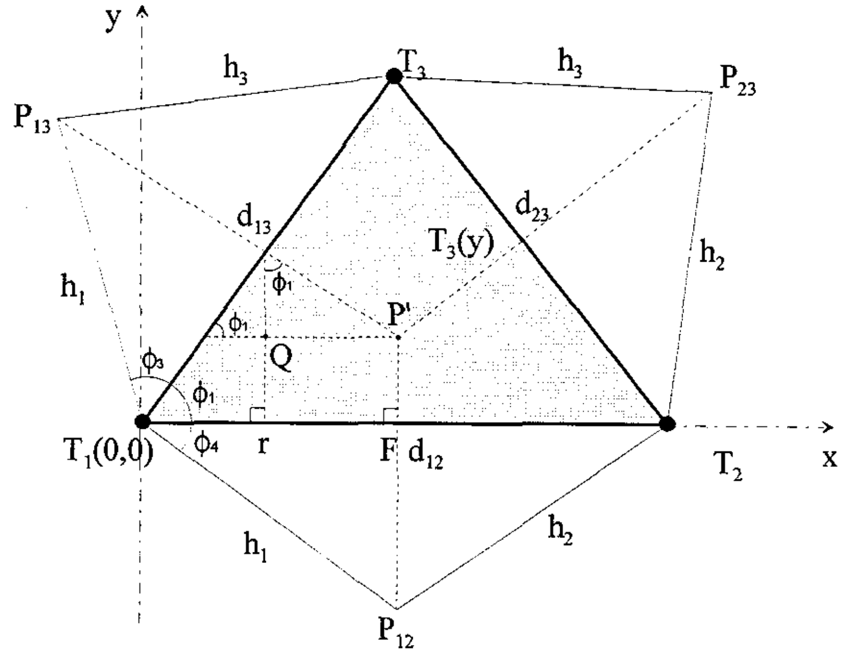


Figure 6.14 Two-dimensional representation

$$\alpha = \frac{\left(h_1^{\,2} + d_{13}^{\,2} - h_3^{\,2}\right)}{2.d_{13}} \qquad (6.37)$$

$$P(x) = \left(\frac{h_1^{\,2} + d_{12}^{\,2} - h_2^{\,2}}{2.d_{12}} \right) \qquad (6.38)$$

$$P(y) = \alpha.\sin\phi - \left(\frac{P(x) - \cos\phi}{\tan\phi} \right) \qquad (6.39)$$

$$P(z) = \sqrt{h_1^{\,2} - P(x)^2 - P(y)^2} \qquad (6.40)$$

The *(x,y,z)* co-ordinate positions can be calculated using equations (6.38), (6.39) & (6.40) and are relative to the transponder plane. The derivation of these equations and the accuracy obtainable is explained in [6.4]. Alternatively, the three-dimensional position of point *P* can be calculated as discussed earlier, using equation (6.23).

6.5 Position Conditioning

There are various steps that are performed when calculating the positions of mobile transponders. An active positioning system starts by measuring the response time of a transponder being interrogated. With prior knowledge of the units' turn-around time the two-way acoustic propagation time can be found by subtracting the constant turn-around time. The unit can calculate the range to the transponder if it has further prior knowledge or the facility to measure the sound velocity of the medium. This can be performed by measuring the temperature, salinity and depth, or by a direct sound velocity measurement using velocity meter. The sound velocity profile often varies, and depends upon the location, time of day, weather and the time of year. In most locations the salinity and pressure are predictable. It is the temperature of the sea as a function of depth that is the most variable and most difficult to predict.[6.3] It is measured by a bathythermograph, which may be deployed from vessels or aircraft, and typically has an accuracy or resolution of ¼ degree Celsius. Using the sound velocity information the range to the transponder can be calculated as indicated in equation (6.1) and Figure 6.1 . Once a minimum of three range measurements have been made the position of the transponder can be calculated in three dimensions. Theoretically there are two possible positions the mobile transponder can be, either in the positive z-domain or the negative z-domain. In practice one of the solutions can be discounted due to the physical positioning of the transponders and valid space in which a mobile transponder can be situated. For example, if the transponders are deployed on the seabed, with the transducer 2 metres above the seafloor, which is relatively flat, the two solutions for the position of the mobile unit are at a depth or altitude of +20m and -20m. It is a fair assumption that the -20m position is incorrect and can be disregarded. Problems occur as the mobile unit approaches the plane of the baseline array, as the position of the mobile unit could be either 1m above the plane or 1m below. This ambiguity can only be removed by having a fourth transponder, preferable situated out of the plane of the other three transponders.

The position of the mobile unit is relative to the transponder baseline array and more importantly the plane of the array, which quite possibly is not parallel to the surface. This can cause a dangerous situation, especially for a diver whom assumes that maintaining a constant altitude corresponds to maintaining a constant depth. The positioning system calculates the altitude of the mobile transponder unit and not the depth. The depth can be calculated, but the depths of the transponders are required so that the position of the diver can be rotated into a plane that is parallel to the surface. Hence, maintaining altitude is the same as maintain depth.

If an absolute position is required, the baseline array co-ordinates may require transposing onto a grid co-ordinate system so that the acoustic positions can be correlated with GPS or previous surveys. If absolute position is not required, the positioning datum maybe a baseline marked by acoustic beacons during an earlier survey. With prior knowledge of the beacons positions during the previous survey the transponder array can transpose and rotate its current coordinates so that they are aligned with the previous survey. The acoustic beacons give the transponder an absolute positioning datum for the local area as GPS gives for the surface of the earth.

Position conditioning is used considerably when using ship mounted SBL or USBL system to compensate for the pitch and roll of the vessel. Using this type of positioning system requires continuous precise monitoring of the ship attitude to enable the measured position coordinates to be transformed into the vessel coordinates. This transformation is best performed in two stages.[6.1] First the linear transformation due to the pitch, θ_p in the *x*, *z* dimensions only. The second transformation in two dimensions of the array measurements into vessel coordinates, involves angular rotation, θ_p. To complete the transformation an angular rotation of θ_r the roll angle must also be performed.

6.5.1 Transposition of Coordinates

The transposing of coordinates is relatively simple and consists of a three-dimensional vector, corresponding to the offset. Consider Figure 6.3 , and assuming beacon 1 represents the positioning datum and the line drawn between beacon 1 and 2 is the grid systems positive x-axis. The position of transponder T_1 can be transposed by subtracting *tx* and *ty* from the position coordinates (x,y,z), as shown in Equation (6.41).

$$\begin{bmatrix} X \\ Y \end{bmatrix} = \begin{bmatrix} x - tx \\ y - ty \end{bmatrix} \qquad (6.41)$$

However, to align the array with the acoustic baseline marker the array also needs to be rotated about the z-axis, indicated by *tr*.

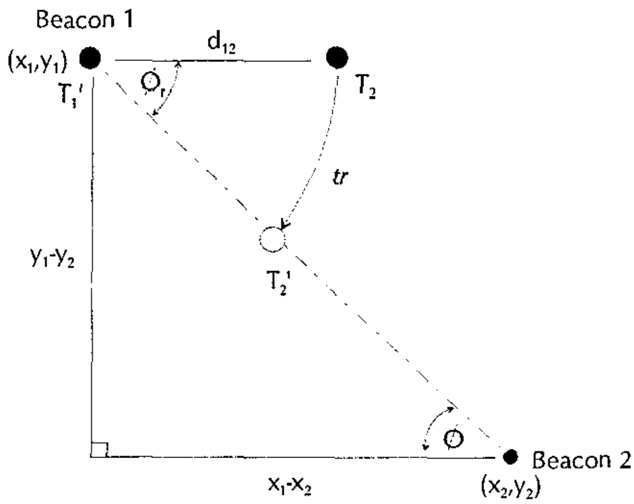


Figure 6.15 Seabed array transformation

Figure 6.15 shows the positions of the transponders after the linear transformation has been performed. By performing the linear transformation the second process of rotating transponder 1 has no effect. Transponder 2 is rotated about (x_1,y_1), which can be assumed to be the positioning datum, not necessarily (0,0). To align the transponder array with the baseline marker beacons requires the coordinates of the transponders to be transposed by the transformation matrix shown in Equation (6.42).

The position of a mobile unit can be calculated relative to the seabed array and then transposed or the coordinates of the transponders can be conditioned, hence transform and rotated so that their positions are referred to the positioning datum. This enables repeated surveys of the same area to be performed with re-deployed transponder arrays, which do not have to be not in the same positions as during the previous surveys.

$$\begin{bmatrix} X \\ Y \\ Z \end{bmatrix} = \begin{bmatrix} \cos\phi & -\sin\phi & 0 \\ \sin\phi & \cos\phi & 0 \\ 0 & 0 & 1 \end{bmatrix} . \begin{bmatrix} x - tx \\ y - ty \\ z \end{bmatrix} \qquad (6.42)$$

Where ϕ is the angle offset between the imaginary lines drawn between beacons B_1 and B_2 and transponders T_1 and T_2.

6.5.2 Three Dimensional Rotation

Figure 6.16 shows the location of the three-seabed transponder relative to transponder T_1, the x,y,z axis shown indicate a x,y plane which is normal to the sea surface.

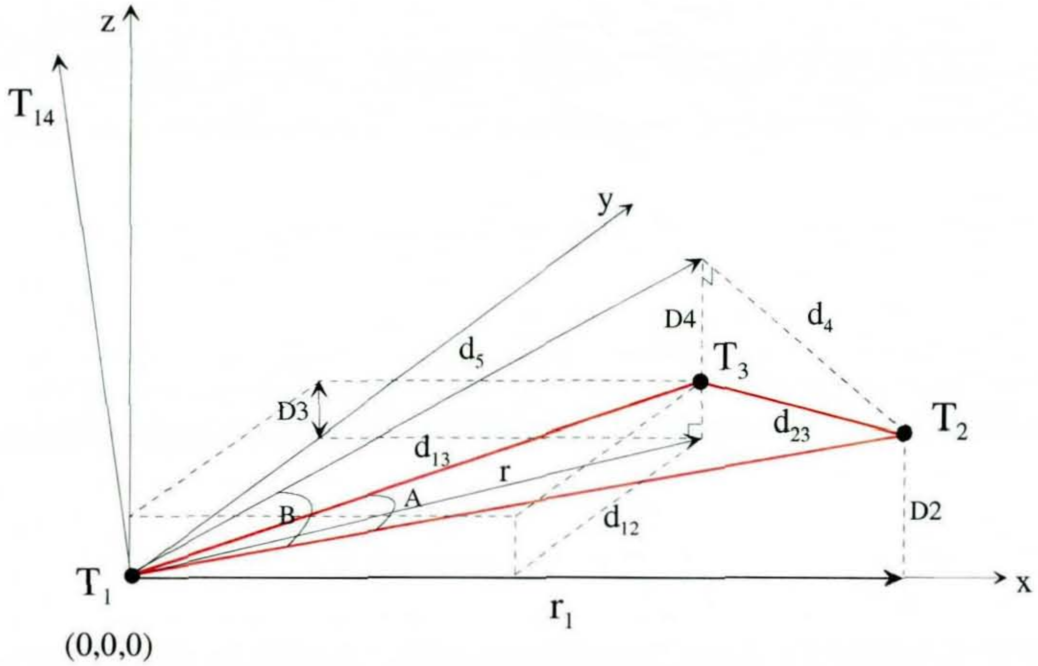


Figure 6.16 Transponder plane in an orthogonal set

To rotate the plane of the transponders into a plane where T_1, T_2 and T_3 are all at the same depth z, T_1 is assumed fixed at depth z. The other two transponders are then rotated about the position T_1 so that their depths are also z. Finding the vector product of T_{12} and T_{13} gives a vector T_{14} acting in a direction perpendicular to the two vector T_{12} and T_{13}, with a magnitude of $T_{12} \bullet T_{13} \sin A$, where A is the angle between the two given vectors. From the resulting vector T_{14} the angles of rotation to eliminate the x and y components of the vector can be calculated.

Vector T_{12} in unit vector terms is:

$$T_{12} = r_1 i + 0j + D_2 k \qquad (6.43)$$

where r_1 is calculated using Pythagoras:

$$r_1 = \sqrt{d_{12}^{\ 2} - D_2^{\ 2}} \quad \text{.....(6.44)}$$

Vector T_{13} in unit vector terms is calculated as follows:

$$D_4 = D_2 - D_3 \quad \text{.....(6.45)}$$

$$r = \sqrt{d_{13}^{\ 2} - D_3^{\ 2}} \quad \text{.....(6.46)}$$

$$d_4 = \sqrt{d_{23}^{\ 2} - D_4^{\ 2}} \quad \text{.....(6.47)}$$

$$d_5 = \sqrt{r^2 + D_2^{\ 2}} \quad \text{.....(6.48)}$$

$$B = \cos^{-1}\left(\frac{d_{12}^{\ 2} + d_5^{\ 2} - d_4^{\ 2}}{2.d_{12}.d_5}\right) \quad \text{.....(6.49)}$$

$$T_{13} = r.\cos Bi + r.\sin Bj + D_3k \quad \text{.....(6.50)}$$

Thus T_{12} X T_{13} yields:

$$T_{14} = r.\sin B.D_2i + r_1.D_3 - r.\cos B.D_2j + r_1.r.\sin Bk \quad \text{.....(6.51)}$$

The resultant vector T_{14} is then rotated by an angle of θ and the resulting vector is then rotated by an angle ϕ. However, as the only known reference is that the vector T_{12} is along the x-axis, the vector must not be rotated about the z-axis. Rotating around the z-axis will shift the *x,y* plane, which can be used when aligning the seabed array with a known datum point. The rotation matrix to rotate the vector T_{14} by θ degrees about the y-axis is as follows:

$$\begin{bmatrix} x_1 \\ y_1 \\ z_1 \end{bmatrix} = \begin{bmatrix} \cos\theta & 0 & -\sin\theta \\ 0 & 1 & 0 \\ \sin\theta & 0 & \cos\theta \end{bmatrix} . \begin{bmatrix} x \\ y \\ z \end{bmatrix} \qquad (6.52)$$

Rotating the resulting vector ϕ degrees about the x-axis gives:

$$\begin{bmatrix} X \\ Y \\ Z \end{bmatrix} = \begin{bmatrix} 1 & 0 & 0 \\ 0 & \cos\phi & -\sin\phi \\ 0 & \sin\phi & \cos\phi \end{bmatrix} . \begin{bmatrix} x_1 \\ y_1 \\ z_1 \end{bmatrix} \qquad (6.53)$$

Multiplying the two-rotation matrices (6.52) and (6.53) together gives a three-dimension rotation matrix (6.54) to allow the diver's position to be rotated into the new co-ordinate set. The diver's position is calculated in the plane of the transponders, then rotated into the new horizontal plane by multiplying the *(x,y,z)* co-ordinates by the matrix (6.54).

The three-dimensional rotation matrix is

$$\begin{bmatrix} X \\ Y \\ Z \end{bmatrix} = \begin{bmatrix} \cos\theta & 0 & -\sin\theta \\ -\sin\phi\sin\theta & \cos\phi & -\sin\phi\cos\theta \\ \cos\phi\sin\theta & \sin\phi & \cos\phi\cos\theta \end{bmatrix} . \begin{bmatrix} x \\ y \\ z \end{bmatrix} \qquad (6.54)$$

The vector T_{14} is evaluated with X and Y equal to zero; hence it has only a Z-component. The angles of rotation θ and ϕ are calculated as shown.

$$\theta = \tan^{-1}\left(\frac{x}{y}\right) \qquad (6.55)$$

$$\phi = \sin^{-1}\left(\frac{y}{r_2}\right)(6.50) \qquad (6.56)$$

$$r_2 = \sqrt{x^2 + y^2 + z^2} \qquad (6.57)$$

Once the diver's position in the new co-ordinate set is found, the position is relayed to the diver and to the surface vessel. If a GPS system is connected to the surface unit the rotation matrix can also include a z-axis rotation to allow the y or x-axis to be aligned with North.

6.6 Conclusion

This chapter shows the overall operation of the system, as to measure ranges and position a mobile transponder all of the components of the system must be operational. This includes the hardware and the software components that facilitate the transfer of data between transponder units.

The practical experiments and simulation outlined in this chapter have shown that the system can perform high-precision range measurements and calculate the position of a mobile transponder to millimetric precision. Positioning accuracy is often difficult to determine, even in controlled condition, but the relative positioning accuracy indicated in Figure 6.8 is ±1mm. This is often more important than true position accuracy as to maintain true position accuracy, channel variation (sound velocity), ray bending and noise will all degrade the true precision.

The relative position accuracy indicates the excellent performance of the hardware and software design. Optimising the algorithms for the platform and considering various Least Mean Squares (LMS) or Kalman filtering algorithms to provide high confidence more reliable position data will improve the overall positioning system.

6.7 References

[6.1] Milne P.H.
Underwater Acoustic Positioning Systems
E. & F.N.Spon, London and New York
1983, ISBN 0-419-12100-5

[6.2] Woodward B. and Goodson A.D.
Diver Tracking by Sonar
Acoustics Letters Vol. 13, No. 2
pp. 25-30, 1989

[6.3] Waite A.D.
Sonar for Practising Engineers
Thomson Marconi Sonar Limited, 2nd Edition, ISBN 0-9528033-1-3
pp. 40, 1998

[6.4] Hodder T.M. and Woodward B.
Algorithms for Underwater Position Fixing
International Journal for Mathematic Education and Science Technology, Vol. 17, No. 4
pp. 407-417, 1986

[6.5] Woodward B. and Zheng Z.
Position Fixing of Transponders from Surface Measurements

[6.6] Woodward B.
Underwater Acoustic Navigation and Tracking Techniques
Acoustics Bulletin, Vol. 25, No. 3
pp. 5-11, May/June 2000

[6.7] Hodder T.M. and Woodward B.
Algorithms for Underwater Position Fixing
International Journal for Mathematic Education and Science Technology, Vol. 17, No. 4
pp. 407-417, 1986

[6.8] Woodward B.
Algorithms for underwater navigation and tracking
3rd European Conference on Underwater Acoustics, Heraklion, Crete, Greece
pp.753 - 758, June 1996

[6.9] Hardman P.A. and Woodward B.
Underwater Location Fixing by a Diver-Operated Acoustic Telemetry System
International Journal on Acoustics 'Acustica' Vol. 55, Research Notes
pp 34-44

[6.10] Wartzok D., Sayegh S., Stone H., Barchak J. and Barnes W.
Acoustic tracking systems for monitoring under-ice movements of polar seals
Journal of the Acoustical Society of America, August 1992 (2) Pt. 1
pp. 682-687, ISSN 0001-4966/92/080682

CHAPTER SEVEN

CONCLUSION AND RECOMMENDATIONS

7. CONCLUSIONS AND RECOMMENDATIONS

7.1 Conclusions

The main aim of this thesis has been to design and develop a real-time underwater acoustic position-fixing system using digital communication techniques suitable for SCUBA divers and underwater vehicles. This is achieved using a novel randomly deployable, self-calibrating array of seabed transponders that forms a long baseline positioning system. The usability of the positioning system is increased by the intelligent protocols, which automatically discover active units and allocate array identification addresses. From the foregoing work and the experimental and simulated results obtained, the following conclusions may be drawn:

1. A new digital underwater communication and positioning system based on the AMD 186 embedded microprocessor, with FLASH memory and SRAM designed into the PCMCIA package, forms the processing module of the system. The design topology is considered in two sections, hardware and software. The design allows implementation of complex digital packet communication and high-resolution time measurements. The software (firmware) is stored in the on-board FLASH enabling in-situation firmware changes and upgrades via the serial interface.

2. The PCMCIA processor module connects to the digital synthesiser and capture module, which can capture signals at 5Msps at a resolution of 12-bits. The synthesiser can generate any waveform and can be transmitted at a rate of up to 1.25Msps at 8-bit resolution. The design allows complex digital modulation and demodulation methods to be used for the generation and reception of acoustic signals.

3. The receiver module is designed to allow the unit to be flexible and was therefore not optimised for a single communication or modulation scheme. The receiver module comprises a low-noise front-end amplifier, an 8^{th} order bandpass Bessel filter (linear phase response) and an Automatic Gain Control (AGC) amplifier stage to

increase the dynamic range of the 12-bit Analogue-to-Digital Converter (ADC). The conditioned output of the AGC is connected to the ADC and the envelope generation circuitry that interrupts the processor module, which causes the processor to capture the signal at the ADC.

4. The linear transmitter module is designed so that any waveform that can be synthesised can be transmitted into the water at a suitable acoustic power level. To design a linear amplifier that maintains linearity up to 100kHz and operates from a low-voltage battery supply is extremely challenging and several compromises had to be made. The push-pull FET power amplifier is designed using a matched pair of power MOSFETs, normally used in audio-applications. The two main problems with FETs are the gate capacitance and the high pinch voltage, when operating from a low supply voltage and at high frequencies. The amplifier is a Class A design, hence when power is connected and there is no input signal present the amplifier consumes a relatively high quiescent current. To prevent this the amplifier supply is switched on 30ms prior to each transmission using a small relay.

5. The acoustic transducers (hydrophones/projectors) used are HS/70 one-inch spheres, with a typical resonant frequency of 75kHz and a Q of 6. This value of Q causes significant problems when attempting to achieve phase changes every two cycles. The matching transformer is not wound to give maximum energy transfer but to reduce the apparent electrical Q. Each transducer is designed and manufactured with a short stubby cable and an eight-way SubConn™ connector. This is so that the transducers can be removed and carefully stored when the system is not in use. Also, removing a transducer gives access to eight connections, which facilitated the downloading of stored data, data upload capabilities, firmware upgrade/changes and battery recharging (of a rechargeable battery pack) without having to open the transponder pod.

6. The mechanical construction of the transponder pod is for controlled shallow water (<35 metres) trials and is not rigorously designed to withstand deployment impacts. The design protects the electronics with a second sealing bulkhead so that if water ingress occurs the battery pack would be destroyed without the electronics being

damaged. The main problem with a plastic housing is the o-ring seating surface, as it is easily scored and can cause the housing to leak under pressure.

7. In all, five complete systems were built, four for testing and one as a backup unit. All of the units are identical and are capable of transmitting and receiving acoustic signals. An expansion port can be used to connect a small graphical LCD screen and Hall effect keyboard to the system. This interface is designed for the mobile diver unit to enable the diver to navigate and input data. Firmware drivers were written for the screen, which includes optimised circle and ellipse drawing algorithms that only use integer mathematics. To draw a circle on the screen requires an ellipse to be calculated to compensate for the aspect ratio of the screen (i.e. the pixel height is greater than its width). Operation and performance of the system was tested in the acoustic test tank at Loughborough University, which is shallow and dominated by multipath interference on long transmissions. During most of the tests the transponders dynamically assigned a Master transponder that discovered the other transponders and all transponders could transmit and receive. The experimental results illustrated that the system performed well in an extremely reverberant environment.

8. To reduce software control during the design and testing of the system, the software for all of the transponder modules is identical. This created an initial overhead in the software design to enable multiple, identical units to be deployed without conflicts that cannot be resolved. Sampling the acoustic noise generated the individuality and this was used as the seed for the random number generator. If no random input is used the units will be synchronised, possibly not in time, but in the random sequences, as the seed in all units will be the same because the software is identical.

The event-driven software design maximises the time that the processor can be In a halt or low-power state, ensuring minimum power consumption. The transponders are capable of measuring the arrival time of a captured signal to a precision of 200ns. Following a packet capture and decode, a precisely timed response is often required to determine range. The transponders are capable of starting to transmit a signal to a precision of 100ns and it was ensured that the software did not degrade the performance through practical testing.

Since the tests were performed in a highly reverberant test tank, long data packets required multiple transmissions of short acoustic packets. The software framing and packet generation layers queue multiple packets on to a timer link-list so that they are transmitted in expiry order. At the receiver the software decoder strips-off the framing and error control bits and re-constructs the information data packet.

9. Various modulation techniques were successfully implemented for the transmission of burst data packets through the underwater channel at a maximum symbol rate of 41k baud. The unique design of the profiled continuous phase modulation scheme used to combat the problems of the high Q transducer proved successful. The design of the continuous phase modulation scheme was implemented to obtain the prescribed high data density communication link between transponders. Simple simulation of normal BPSK and QPSK modulation schemes with additive white Gaussian noise proved unrealisable due to the hardware and transducer characteristics.

   Binary phase shift keying with a 90° phase change to indicate a data bit change (differentially encoded) was implemented and proved successful. It was also found that a phase retard rather than a phase advance produced less amplitude fluctuation in the received waveform, which is due to the characteristics of the hydrophone. The result is a rotating clockwise vector, which changes phase a maximum of 90° per symbol. Although not originally designed to be Offset PSK, the resultant modulation scheme is very similar, which led to the development of the Continuous Phase Offset Quadrature Phase Shift Keying (CPOQPSK) scheme. The symbol period was initially extended to twice the CPBPSK scheme and hence the data rate remained constant. The data rate was increased by reducing the symbol period from four to three cycles, thereby increasing the data rate by 25%. This is a significant saving as the longest standard packet length is reduced to 1.26ms, a saving of 400μs.

The decoding of a data packet is post-processed following capture and the BER performance for the various techniques is plotted. The BER tests were performed initially by interconnecting the two systems, transmitter and receiver, using the RS232 port, however as the systems only have one asynchronous serial port debug information could not be generated using this configuration. To overcome this the two systems synchronised their random number generator seed at start-up, hence the transmitter cycled through and transmitted the same sequence of numbers as the receiver was expecting. Tight timing constraints ensured that if the receiver missed a transmission the system would remain synchronised.

10. Positioning algorithms in an active system are based on the intersection of spheres, with a unambiguous solution found if there are at least four known ranges. The positioning algorithms are well documented and an implementation of these algorithms is shown. MatLab simulations of the three-dimensional algorithms indicate the precision of the algorithms. The standard floating point C math library enables the AMD186 processor to calculate mathematically intense positional information. The processor module is capable of calculating the position of the unit in less than a 500ms.

11. The low-power constant source level acoustic beacon discussed can be used in a variety of underwater acoustic application. The original design, PICE™ (Porpoise Incidental Catch Eliminator) is used for marking fishing nets to prevent the by-catch of cetaceans. Other applications include: acoustic marking of transported cargo, diver shot-line marker, acoustic reference source, long-term point-of-interest marker and reference markers to enable a seabed array to be re-aligned with a previous survey as discussed. The standard commercially available beacon used in the fisheries has an active life of approximately 2 years. The beacon produces a constant source level of 145dB re 1μPa throughout its life and is powered by a single D Cell battery. The University has patented this acoustic device and the fisheries variant is manufactured under licence by a UK based company. The diver shot rope version has been under development with a commercial diving company collaborating with the University. The acoustic reference source was design for calibration purposes in the field, as the device can be calibrated and due to it constant source level throughout its battery life, the unit can act as a reference source. The use of inexpensive disposable beacons for baseline marking that enables the array to mathematically re-align itself with an earlier positioning tasks can offer significant time and cost savings. This concept eliminates the need for repeated geodetic

calibration tasks to be performed, as long as the seabed beacons are still functioning and have not moved.

12. The next version of the acoustic beacon is an interactive design, hence a miniature transponder, which was originally design before the beacon mode device. This unit proved uneconomic for the fishing industry, however for acoustic reference marking, an inexpensive transponder device would be extremely useful. A prototype dual frequency interactive beacon was demonstrated in the test five years ago, which lead to the development of the now patented PICE™ device.

7.2 Recommendations

The underwater position fixing and communication system presented in this thesis is a prototype. The system has hardware and software limitations and it is suggested that they should be taken into consideration in future designs. In this section, these limitations are introduced and improvement methods are discussed.

7.2.1 Hardware

An important requirement for an underwater system, which is to be used by divers, is that it must be small and unobtrusive and easy to use. The graphical interface is relatively small, however it was not encapsulated for underwater use during this project. The intention was to have the screen with the electronics and battery pack mounted on the wrist like a dive computer. This seems to be an unrealistic approach unless the electronics can be further miniaturised and the power consumption reduced. Since the system is being used in an underwater environment the display should not be damaged by high ambient pressure. To achieve these requirements, an ergonomic design of the system is necessary and suitable packaging must be investigated.

The system was design in modules: processor and memory; digital capture and synthesiser; receiver and power transmitter. To reduce the size, inter-board connection and reliability, these four boards should be designed onto a Printed Circuit Board (PCB).

To reduce power consumption the new AMD186ER processor should be used. This processor is based on the EM/ES version used and offers new features, such as 32K of on-chip SRAM and 3.3V operation with 5V tolerant inputs. With careful design the 32K of SRAM should be sufficient for the transponders. The changes above will possibly give a 50% processor module power saving. Further power savings can be achieved by reducing the processor clock from 40MHz to 20MHz, which also reduces the ADC speed requirements. This will degrade the overall timing accuracy of the system and increase position calculation times.

Alternatively, the designer could look towards a complete System-On-Chip (SOC) solution, as the cost and size of FPGAs is encouraging this type of approach. This would be a more expensive approach initially, however for a commercial miniaturised, low-power and reliable system this is possibly the way forward.

The receiver module should be re-design for the specific communication technique employed and the frequency band of interest. The filter on the receiver should be replaced with a digitally configurable bandpass filter, such as the Zetex ZXF103. This will enable the system to operate on processor controlled frequency band and all though off the resonant frequency of the transducer, if the range to a neighbouring transponder is short then high frequency communication would be advantageous. The advantage of a multiple frequency band system with dynamic, processor controlled band detection is a natural development of an acoustic network. The possibilities that multiple frequency band operation offers include; array isolation, dedicated communication channels, contention based network access channel, dedicating positioning frequency channel.

The interrupt that is generated by the envelope of the filtered signal should be improved to prevent false triggering, which causes the system to miss data packets and increases current consumption.

To improve the receiver module the AGC circuit should be eliminated or replaced with a digitally controlled amplifier gain circuit, as it is difficult to design an AGC circuit that has a linear phase response throughout its dynamic range. A digitally controlled amplifier is still not ideal for a multi-point communication network as the amplification level for the capture packet can only be determined after the data packet has been captured. In a point-to-point communication system the amplifier gain can be adjusted to ensure that the acoustic signal is sufficiently amplified and set to an optimal level after two or three acoustic exchanges. However, with the system described this is not possible as the acoustic intensity of the received signal will be dependant on the range to the transmitting transponder unit.

Changing the 12-bit ADC to a 16-bit ADC will increase the dynamic range of the receiver by 26dB. It is essential to have a constant or processor controllable input level when digital filtering and channel equalisation.

The design of the Class A output power amplifier should be changed to a Class B or C to improve power consumption and eliminate the need to switch the supply, which causes acoustic impulses to be transmitted when the power supply is switched.

The transponder housing should be machined from aluminium and made larger to accommodate a bigger battery pack. The transducer should be moulded onto the topside of the transponder and protected by a guard, which does not significantly affect the beam pattern of the transducer. So that the acoustic power output of the transponder and the communication-symboling rate could be increased and the transducer used should be investigated, as a spherical projector is not required on the seabed units. A hemi-spherical beam pattern would be ideal; as at present energy transmitted down into the seabed is wasted and increases multipath interference problems during both transmit and receive modes. Retrieval of the seabed units should be possible by sending an acoustic release command, which instructs the transponder to drive a release mechanism.

7.2.2 Design and Software

In the design of underwater communications the phase modulation techniques used proved to function well in the tank tests, however the true performance can only be assessed during a real position-fixing task. The coherent phase modulation technique requires a synchronisation header that with further investigation can be used for Doppler or packet compression/expansion correction during decoding of the data packet. Also, on multi-user systems the phase modulation can be applied to different carrier frequencies, allowing multiple transponder arrays to be used side-by-side.

The accuracy of the system is determined by the detection of the Barker code at the head of the packet, which also provides symbol synchronisation. Barker codes were used as headers to the communication packets as they are the optimum code, which gives an unambiguous correlation peak. However, as the SNR decreases the accuracy of detection will also decrease. To counteract this effect the code length can be increased and it is suggested that the researcher investigates the improvement in accuracy with respect to Barker code length and SNR.

The multipath propagation phenomenon of underwater communication channels introduces serious limitations in signal decoding and it is recommended that further investigation should be made into multipath elimination methods, especially for the seabed transponder that are stationary. A multi-channel equalisation scheme was deigned into the protocol for the static seabed transponders, to reduce the multipath interference from the seabed when the units are widely spaced. In point-to-point communication systems the impulse response of the channel is measured and equaliser tap weights calculated. Whereas, in a navigation system each transponder has unique channel, hence a different equaliser. This presents a problem, as the receiving unit needs to know the source of the transmission to be able to use the correct equalisation filter. The post-processing of the data packet following capture enables the demodulator to extract the *from_id* from the captured data packet. This allows the correct equalisation process to be performed on the data packet, as the source of the transmission is known. Hence the limitation of this design is that the direct path must arrive at least 250μs before the first multipath. Further development of channel equalisation methods in general will enable the performance of the system to be improved dramatically. The design of communication packet with training sequences imbedded to enable adaptive equalisation algorithms to be used in none static situations.

The accuracy of the system was calibrated using control tests in the acoustic test tank and the three-dimensional position-fixing algorithms were simulated to prove the feasibility of the overall positioning system. However, an open water positioning exercise with divers and a deployed array would be interesting, however this requires experienced diver's and other expensive equipment to accurately confirm the diver's position. The biggest problem with an open water trial is proving that the diver is where the positioning system indicates he is!

The baseline marker pingers require programming with suitable codes as discussed and acoustic tests to prove the attainable accuracy that the system can passively position a beacon. If the marker pingers can be accurately positioned during acoustic test the *protocols and positioning grid translation mathematics can be incorporated into* the design. The software can be easily upgrade to accept passive positional data, as it would be included as a passive positioning state.

The network of transponder on the seabed could be connected to other logging equipment, which requires a medium speed (>1kbits/s) data communication link to the surface vessel. The vertical data communication link would require investigating and suitable error correction coding, channel equalisation and differential coding techniques.

The position fixing algorithms can be improved significantly by optimising the algorithms for the platform and calculating the position using LMS or RMS algorithms. Further investigation should be made into Kalman filters to improve the performance when acoustic dropout occurs.

Last but not least, the digital underwater position system designed and developed throughout this study has overcome some very interesting practical problems. Several of the novel aspects outlined in this thesis are being implemented in future commercial acoustic positioning systems that are currently under development...

THE END

Publications

8. PUBLICATIONS

Journal Papers - Academic Journals

1. Zheng, Z., Woodward, B. and **Newborough, D.** "Calculating the allowable co-ordinates of acoustic transducers to avoid multipath interference in a tank", ACUSTICA - Acta Acustica 2002 (in print).

2. **Newborough, D.**, Blomqvist, C., Lepper, P.A. and Goodson A.D. "The new generation of Electronic Click Detector (ECD): development and field trials data", *Proceeding of the Institute of Acoustics*, Volume 23, Part 4, 23-24 July 2001, pp 187-198, ISSN: 0309 8117.

3. **Newborough, D.**, Goodson, A.D. and Woodward, B., "An Acoustic Beacon to Reduce the By-catch of Cetaceans in Fishing Nets", *Journal of the Society for Underwater Technology*, Summer 2000, Vol. 24, Number 3., pp 105-115, ISSN: 0141 0814.

4. **Newborough, D.**, Goodson, A.D. and Woodward, B., "Micro-Controller Based Deterrents: Acoustic Devices to Reduce Harbour Porpoise (*Phocoena Phocoena*) Incidental Catch in Gillnets", *Proceeding of the Institute of Acoustics*, Volume 19, Part 9, December 1997, pp 235-243, ISSN: 0309 8117.

5. Dudzinski, K. M and **Newborough, D.**, "Concurrent recording of dolphin behaviours, frequency-modulated tones, and pulsed vocalistions (including echolocation clicks) underwater with a swimmer-propelled system", *Proceedings of the Institute of Acoustics*, 19(9), December 1997, pp 199-207, ISSN: 0309 8117.

Journal Letters

1. Connelly, P., Woodward, B., Goodson, A.D., Lepper, P. and **Newborough, D.**, "Remote Sensing Methods for Cetacean Interactions with Pelagic Trawl Fishing Gear", *11th Annual Conference of the European Cetacean Society (Abstract)*, March 1997, pp 17.

2. Amundin, M., Blomqvist, C., Larsen, F., Lepper, P.A., Lockyer, C.H., Goodson, A.D., Mayo, R.H. and **Newborough, D.**, "The Initial Reaction of Two Wild Harbour Porpoises to an Innovative Wide-Band Acoustic Gillnet Deterrent", *Abstracts of the 25th Annual Symposium of the European Association for Aquatic Mammals (Abstract)*, March 1997, pp 22.

3. **Newborough, D.**, Goodson, A.D. and Woodward, B., "Beacon Mode Deterrents for Gillnets" , *11th Annual Conference of the European Cetacean Society (Abstracts)*, March 1997, pp 41.

Conference proceedings

1. Woodward, B. and **Newborough, D.,** "Information technology advances in underwater tracking and communications", UDT Pacific 2000, Sydney, Australia, February 2000, pp 134 - 136.

2. **Newborough, D.** and Woodward, B., "Diver Navigation and Tracking System", Oceans'99 MTS/IEEE Conference, Seattle, September 1999, Vol. 3, pp 1581 - 1586, ISBN 0-933957-24-6.

3. Goodson, A.D., **Newborough, D.** and Woodward, B., "Set Gillnets Acoustic Deterrents for Harbour Porpoises, *Phocoena phocoena*: Improving the Technology", *ICES Annual Science Conference 1997*, 85th Statutory Meeting, Baltimore, USA, October 1997, pp. 1.

4. Goodson, A.D., Amundin, M., Mayo, R.H., **Newborough, D.**, Lepper, P.A., Lockyer, C., Larsen, F. and Blomquist, C., "Aversive sounds and sound pressure levels for the harbour porpoise (*Phocoena Phocoena*): An initial field study", *Annual Science Conference of the International Council for the Exploration of the Sea* , Baltimore, USA, 1997, pp. 5.

5. Mayo, R.H., Amundin, M., Goodson, A.D., Lockyer, C.M., Lepper, P.A., **Newborough, D.**, Larsen, F. and Blomquist, C., "Observed surfacing behaviour of wild harbour porpoises" , *11th Annual Conference of the European Cetacean Society* , Stralsund, Germany, In:*European Research on Cetaceans - 11* P.G.H. Evans, E.C.M. Parsons and S.L. Clark (eds), 1997, pp. 4.

6. Goodson, A.D. and **Newborough, D.**, "Designing Efficient Pingers or Fishing Nets: Aversive Sound Thresholds for Harbour Porpoises", *Abstracts of the 25th Annual Symposium of the European Association for Aquatic Mammals (Abstract)*, March 1997, pp 12.

7. Goodson, A.D., **Newborough, D.** and Woodward, B., "Interactive Deterrent Devices for Fishing Nets Designed to Reduce Small Cetacean Bycatch", *International BioAcoustics Council XV IBAC Symposium,* G. Pavan, University of Pavia (Italy), Pavia, University of Pavia, Italy, October 1996, p 32.

Other Publications - Research Equivalent

1. **Newborough, D.**, Goodson, A.D. and Woodward, B., "By-Catch Reduction Acoustic Device", International Patent Classification Application No PCT/GB97/01976, AOIK 79/02, International Publication No WO 98/03062, 1998, 1p.

2. Goodson, A.D., **Newborough, D.** and Woodward, B., "By-Catch Reduction Acoustic Device", United States Patent No.: US 6,170,436, 9th January 2001.
"By-catch reduction", British Patent application, No. 9615237.6, 19 July 1996; "Acoustic device", British Patent application, No. 9616559.2, 6 August 1996; "By-catch reduction acoustic device", European Patent application, No. 97932925.7.

3. **Newborough, D.**, Goodson, A.D. and Woodward, B., UK Trademark No. 2165305, Loughborough University, 30 April 1998, "PICE", Porpoise Incidental Catch Eliminator and Dolphin Device.

9. APPENDICES

APPENDIX A

SYSTEM DESIGNS

APPENDIX A

SYSTEM DESIGNS

1. Appendix A: System Designs

1.1 Contents

PCB Layout Design

Input Circuitry

Analogue to Digital Converter design

Digital to Analogue Converter Design

Input Output Select Relay

Power Supply

Graphical Interface

Hall Effect Input Switches (1 of 2)

Input Switches (2 of 2)

IR Input Switches

AMD Processor Module Drawing List

AMD Board Design

PCMCIA Interface Connections

Processor Board Pull-ups

ES/EM Power Decoupling

RS232 interface

Processor Board FLASH

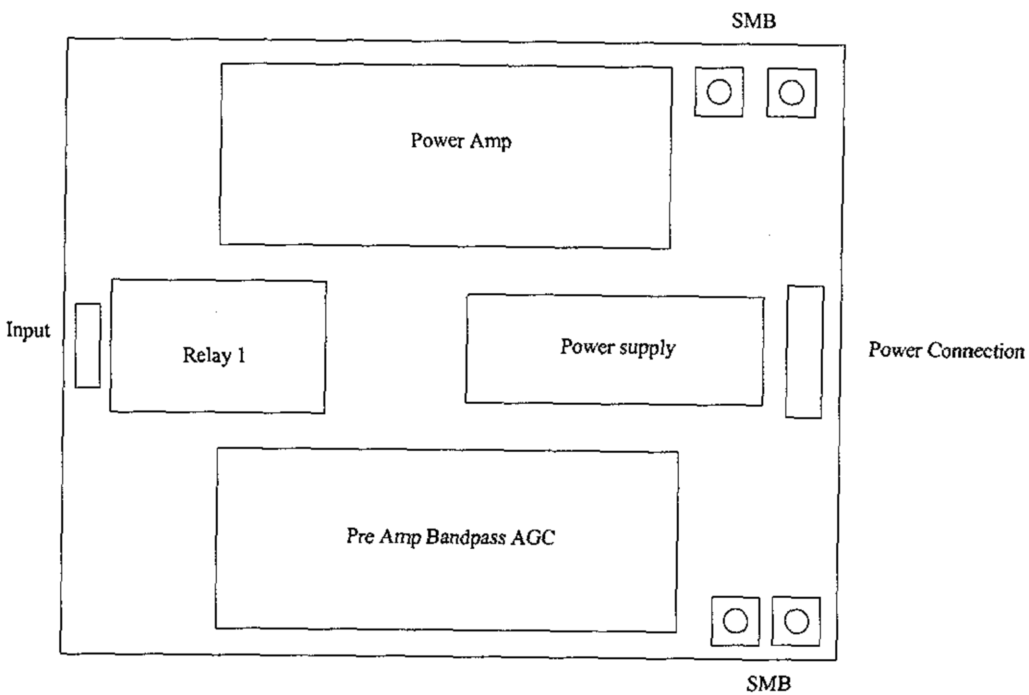
Processor Board SRAM

Processor Board Regulators

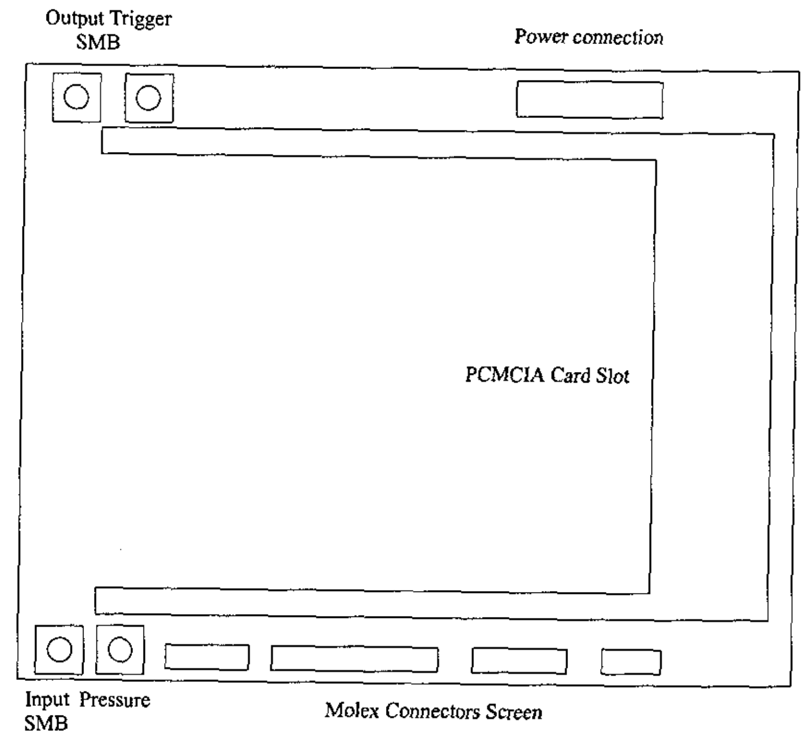
Processor Board Reset Module

AMD Processor Board 5V version Component List

Analogue Board



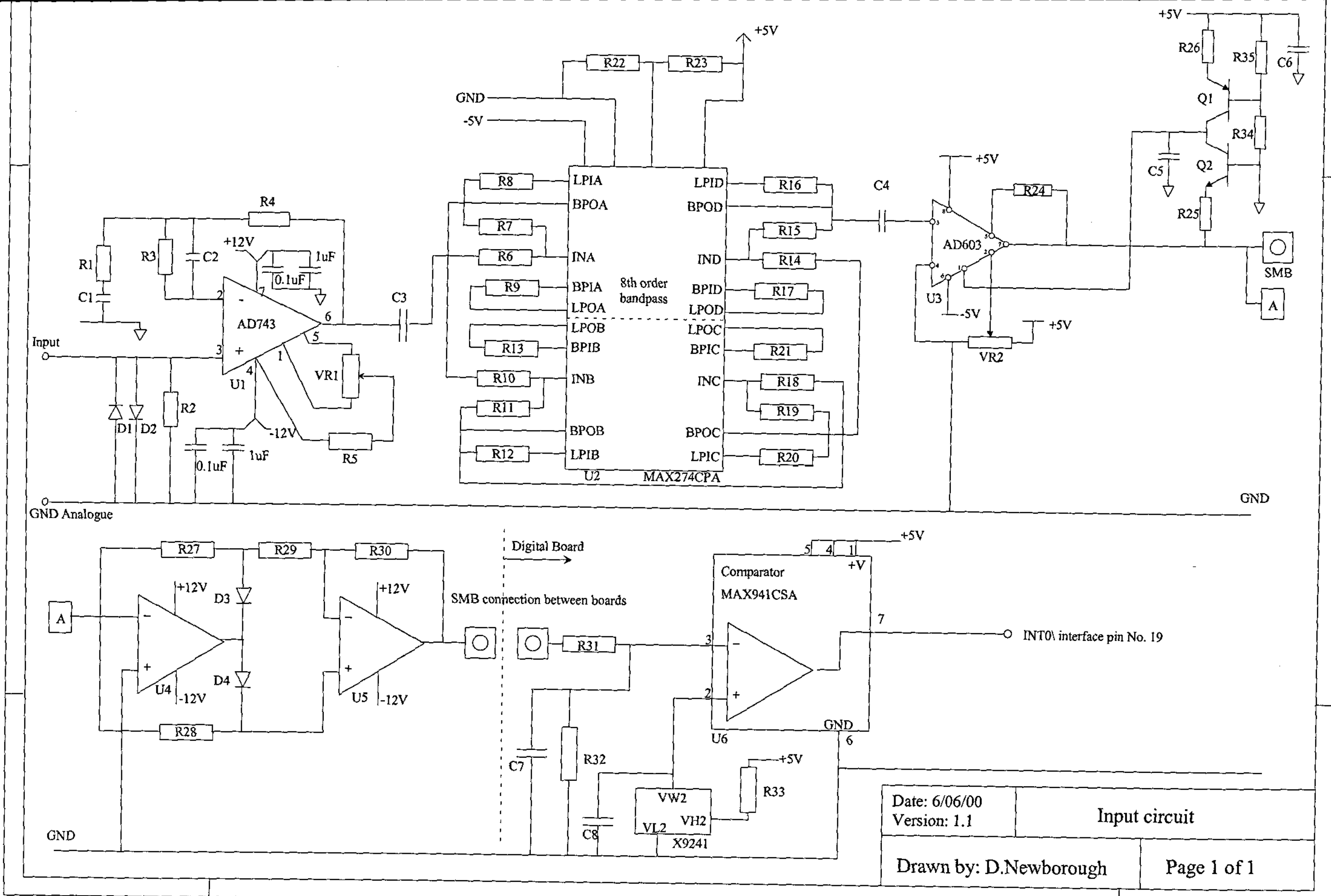
Digital Board



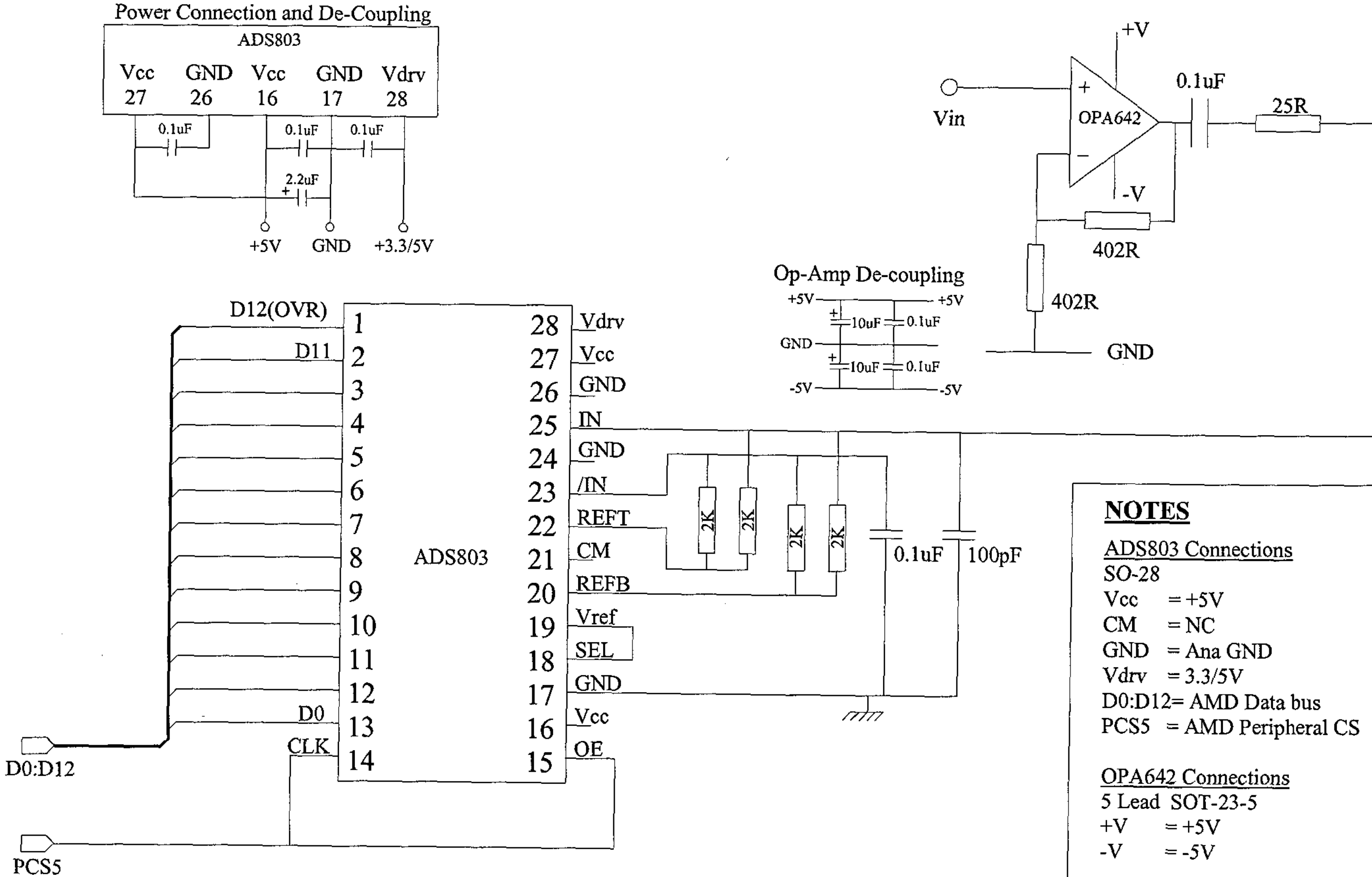
NOTE:

Layout is only advisory

Date: 08/06/00 Version: 2.0	Drawn by: Darryl Newborough
Title: PCB Layout design	



Input
GND Analogue
R1
C1
R3
C2
R4
+12V
1uF
0.1uF
AD743
U1
VR1
R5
-12V
0.1uF
1uF
D1
D2
R2
C3
GND
-5V
R22
R23
+5V
R8
R7
R6
R9
R13
R10
R11
R12
LPIA
BPOA
INA
BPIA
LPOA
LPOB
BPIB
INB
BPOB
LPIB
8th order
bandpass
LPID
BPOD
IND
BPID
LPOD
LPOC
BPIC
INC
BPOC
LPIC
R16
R15
R14
R17
R21
R18
R19
R20
U2
MAX274CPA
C4
AD603
U3
R24
VR2
+5V
-5V
R26
R35
C6
Q1
R34
C5
Q2
R25
SMB
A
GND
R27
R29
R30
A
+12V
-12V
D3
D4
U4
U5
R28
Digital Board
SMB connection between boards
R31
C7
R32
C8
VW2
VH2
VL2
X9241
R33
Comparator
MAX941CSA
+V
U6
GND
INT0\ interface pin No. 19
Date: 6/06/00
Version: 1.1
Input circuit
Drawn by: D.Newborough
Page 1 of 1



NOTES

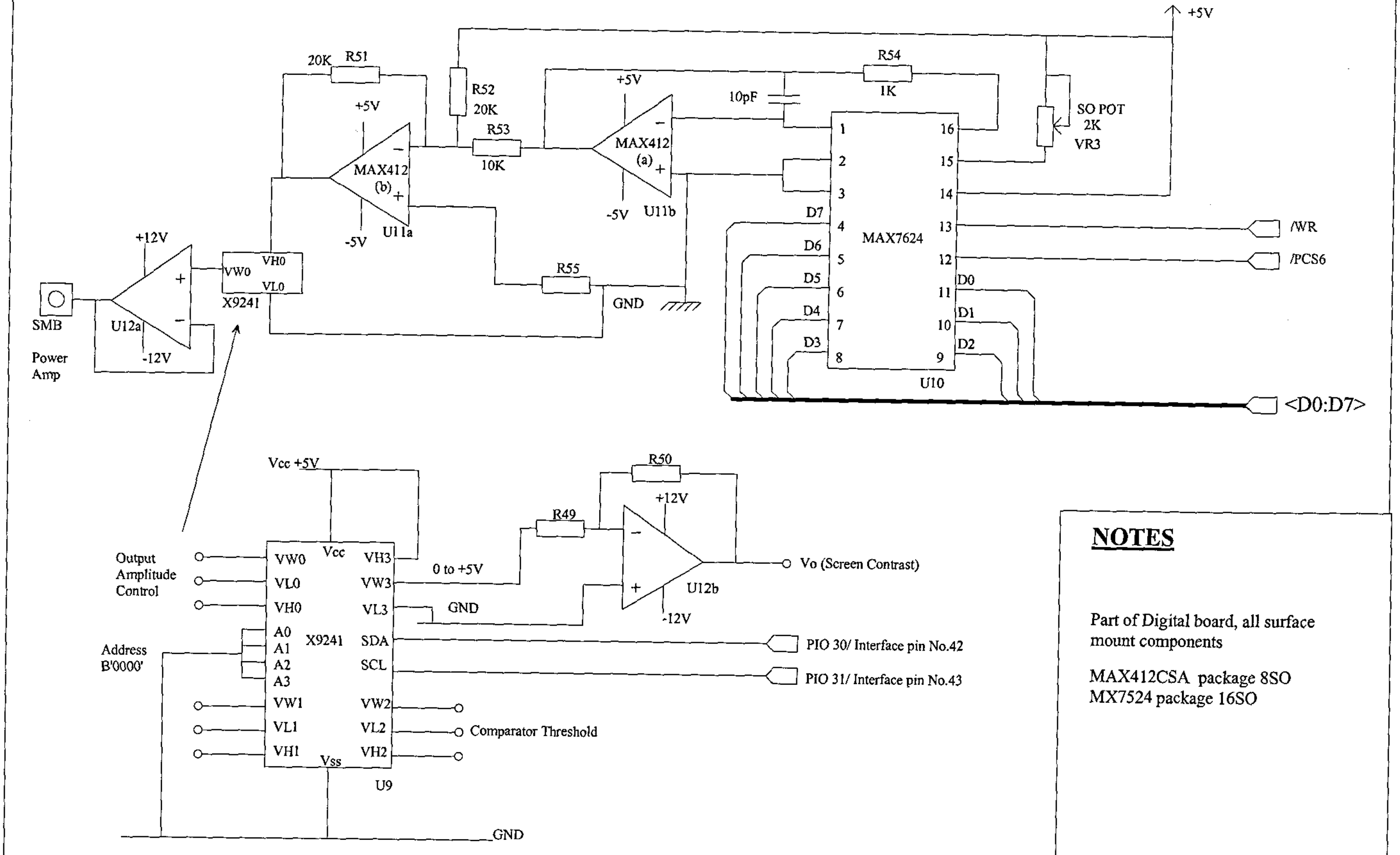
ADS803 Connections
SO-28
Vcc = +5V
CM = NC
GND = Ana GND
Vdrv = 3.3/5V
D0:D12= AMD Data bus
PCS5 = AMD Peripheral CS

OPA642 Connections
5 Lead SOT-23-5
+V = +5V
-V = -5V

Date: 04/04/00

Title: Analogue to Digital Converter Design

Drawn by: Darryl Newborough



NOTES

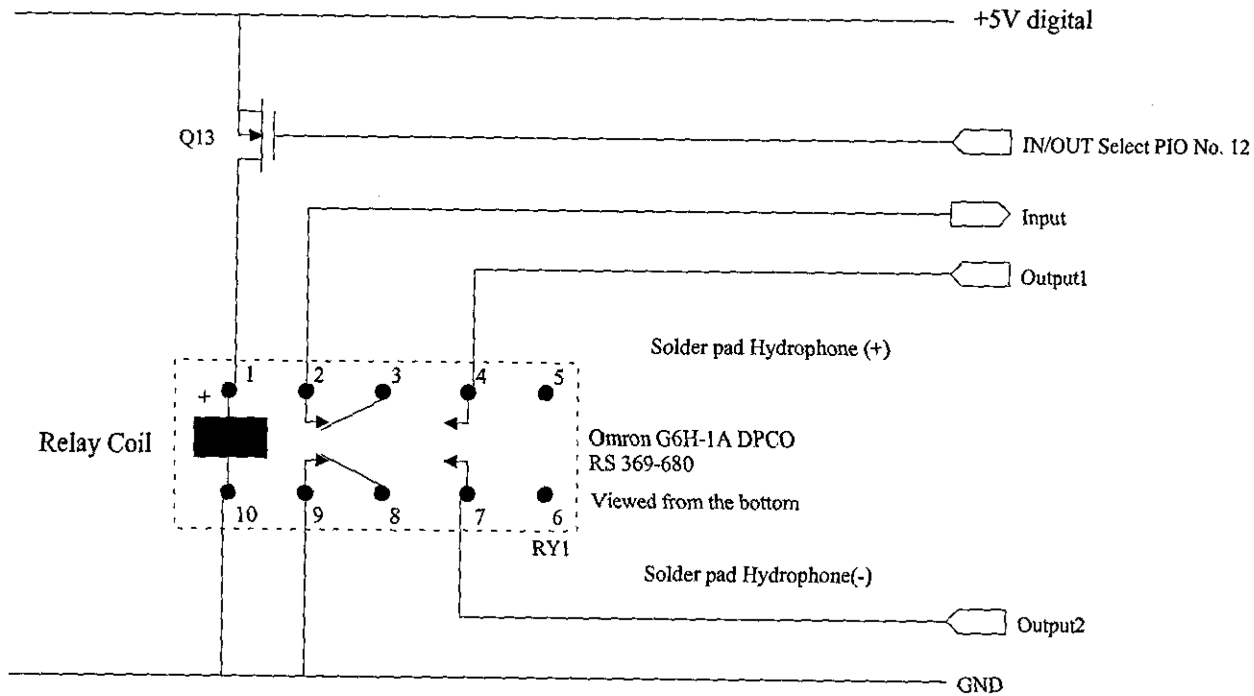
Part of Digital board, all surface mount components

MAX412CSA package 8SO
MX7524 package 16SO

Date: 04/04/00

Title: Digital to Analogue Converter Design

Drawn by: Darryl Newborough



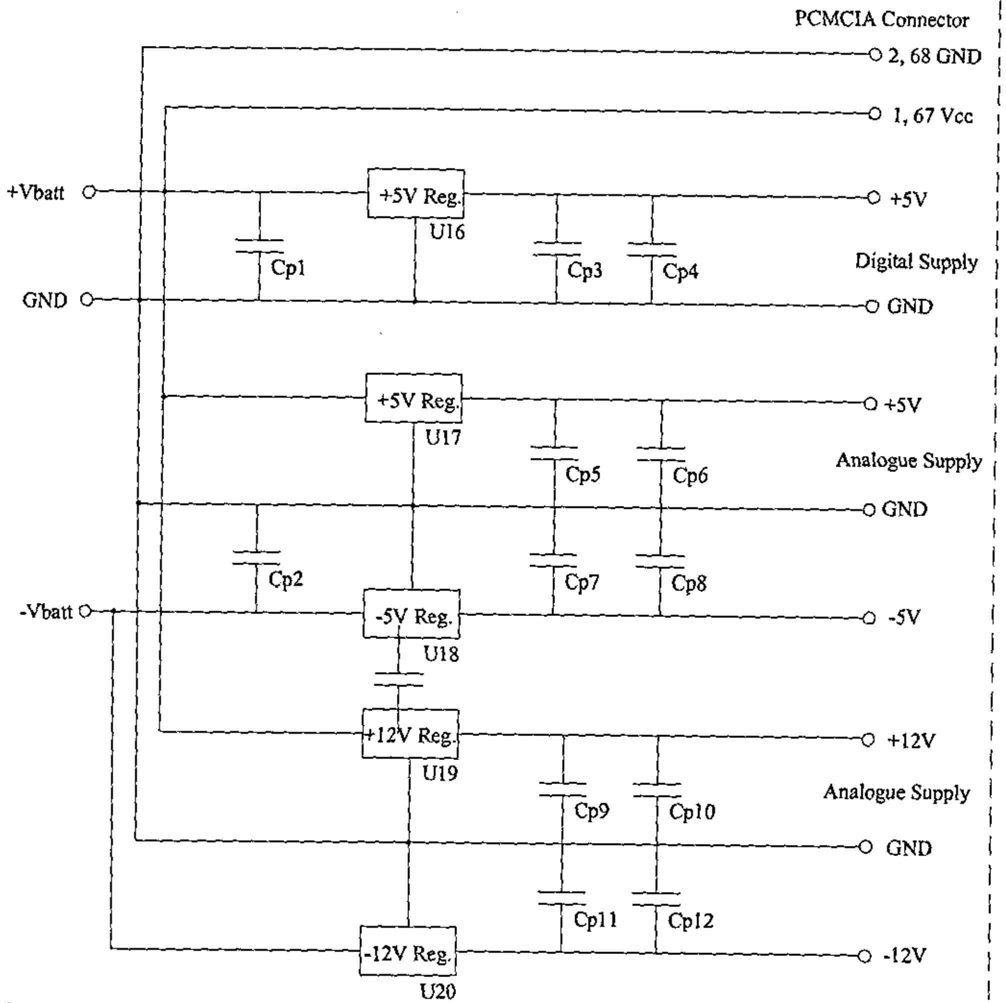
NOTE:

Date: 22/05/00
Version: 2.0

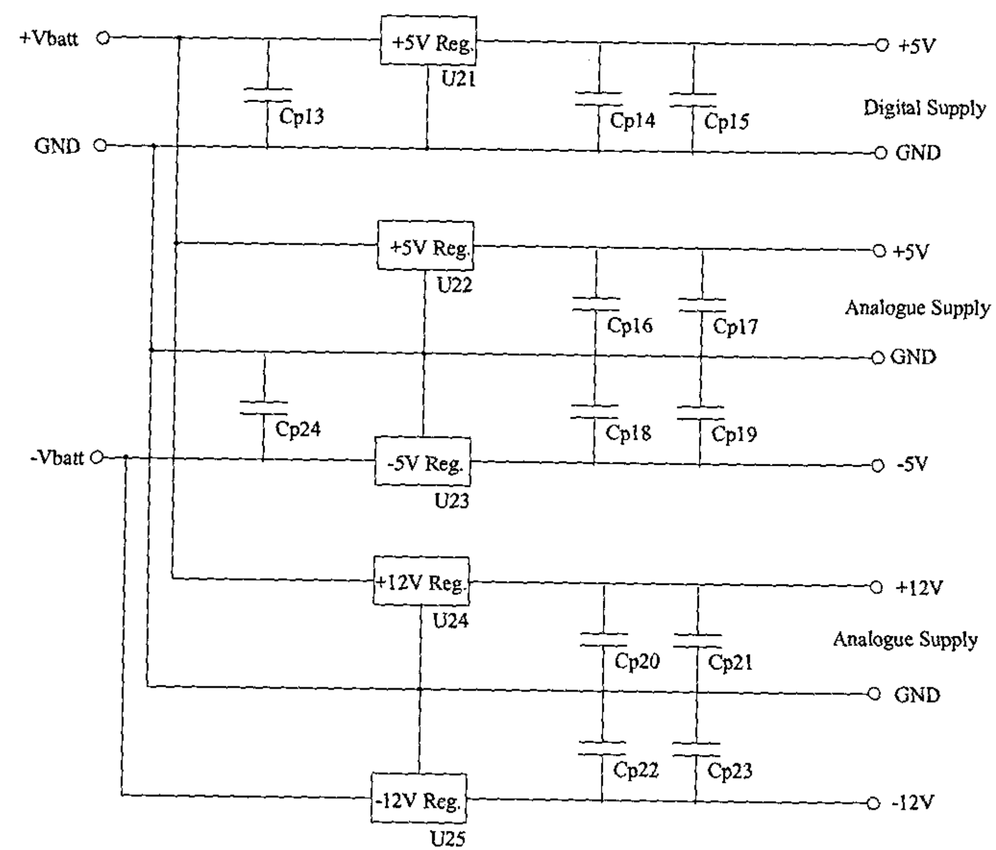
Drawn by: Darryl Newborough

Title: IN / OUT Select Relay

Digital Board Power Supply Setup

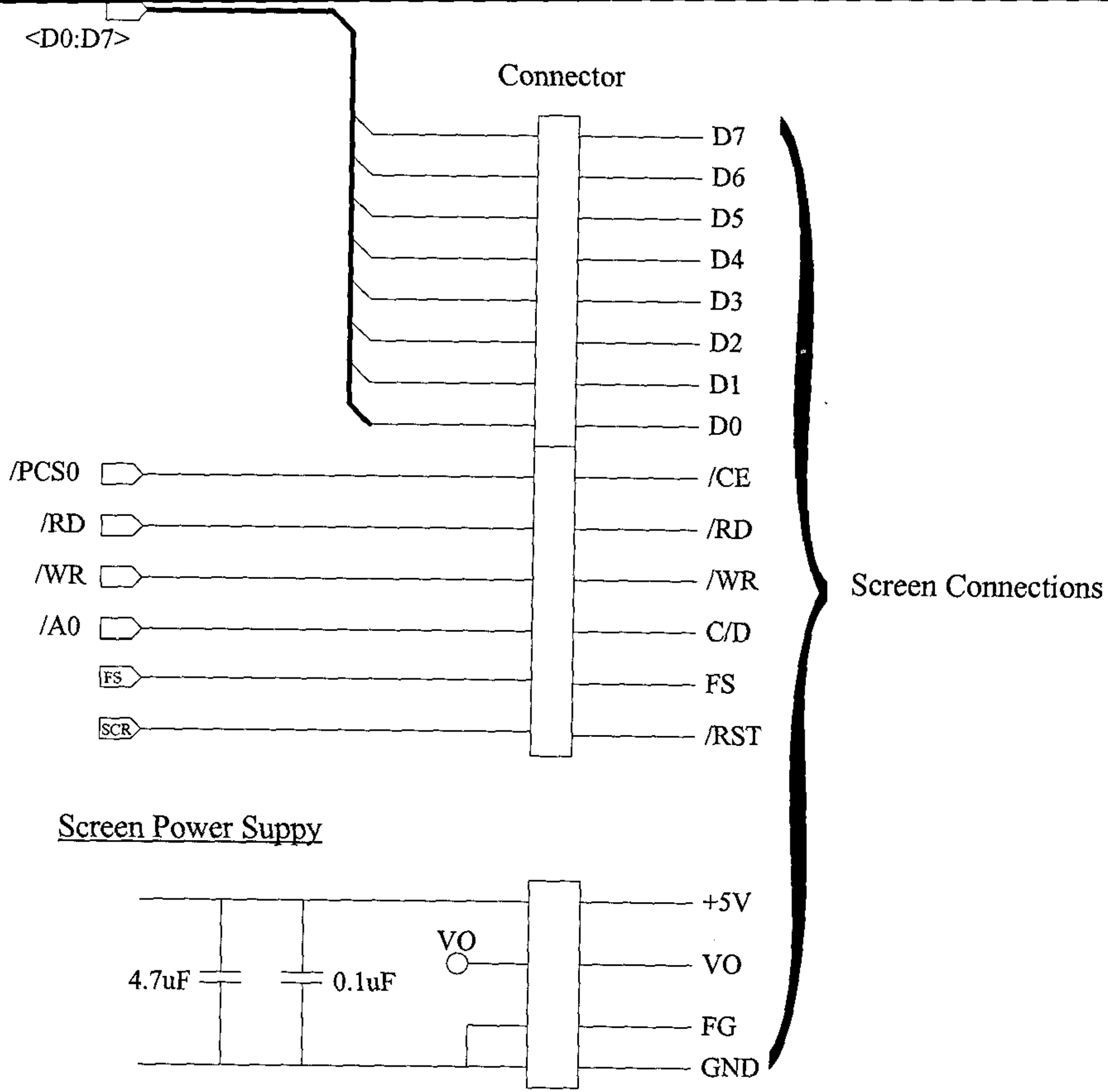


Analogue Board Power Supply Setup



NOTE:

Date: 08/06/00 Version: 1.0	Drawn by: Darryl Newborough
Title: Power Supply	



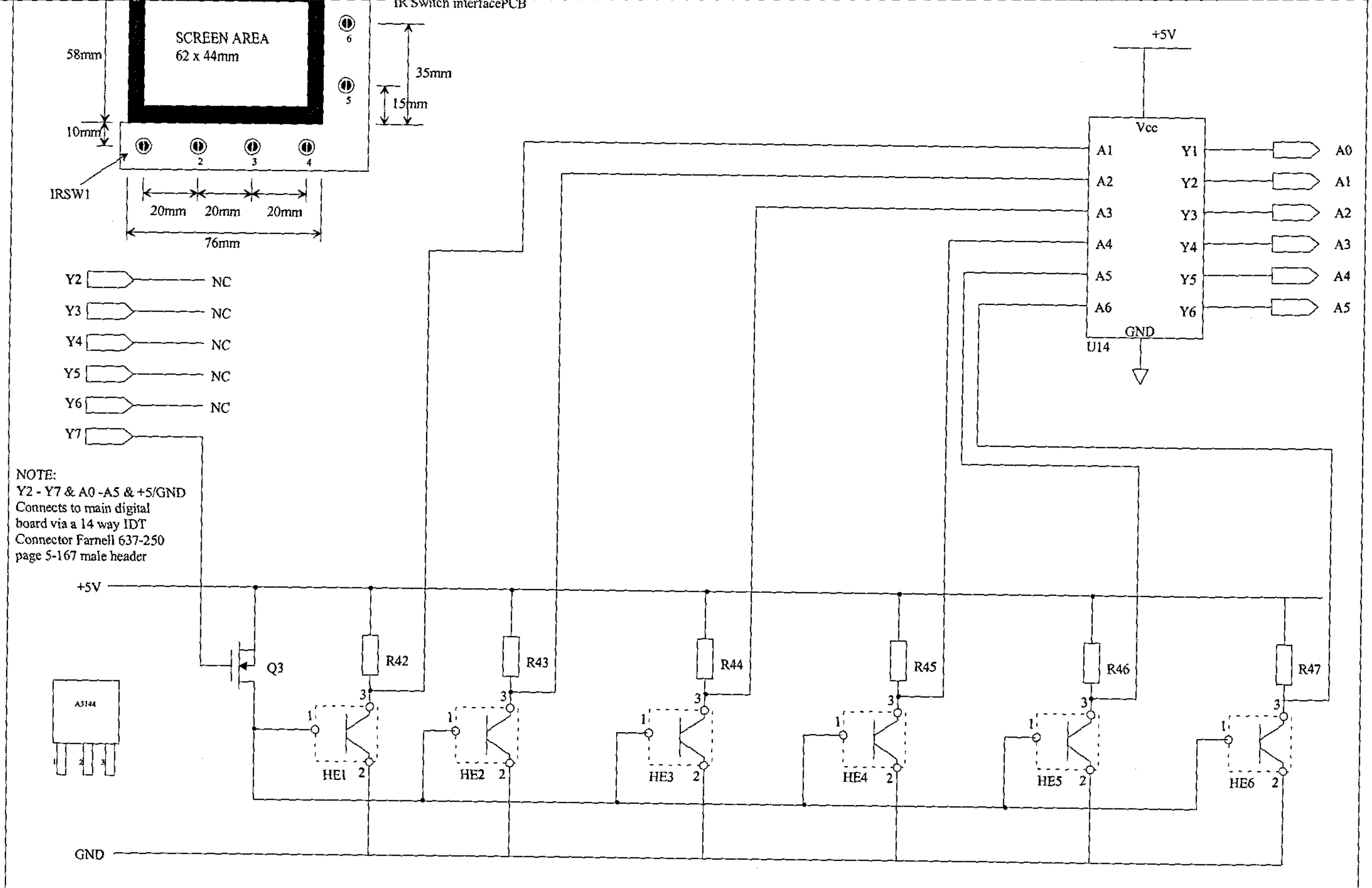
NOTES

SCR	= PIO 1
FS	= PIO 2
C1	= PIO 3
C2	= PIO 4
C3	= PIO 5

Date: 04/04/00

Title: Graphical Interface

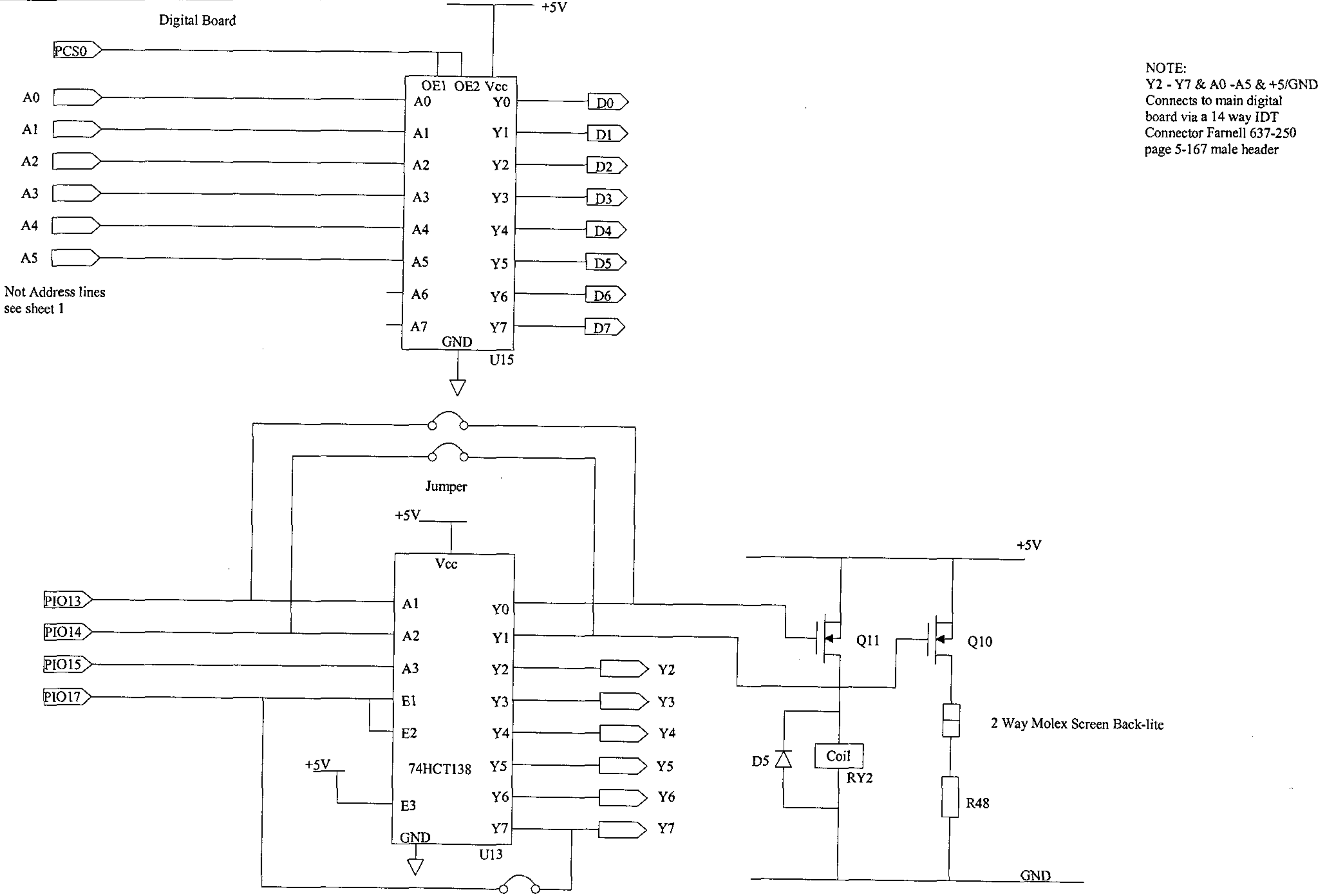
Drawn by: Darryl Newborough



Date: 06/06/00 Page 1 of 2

Hall effect Input Switches

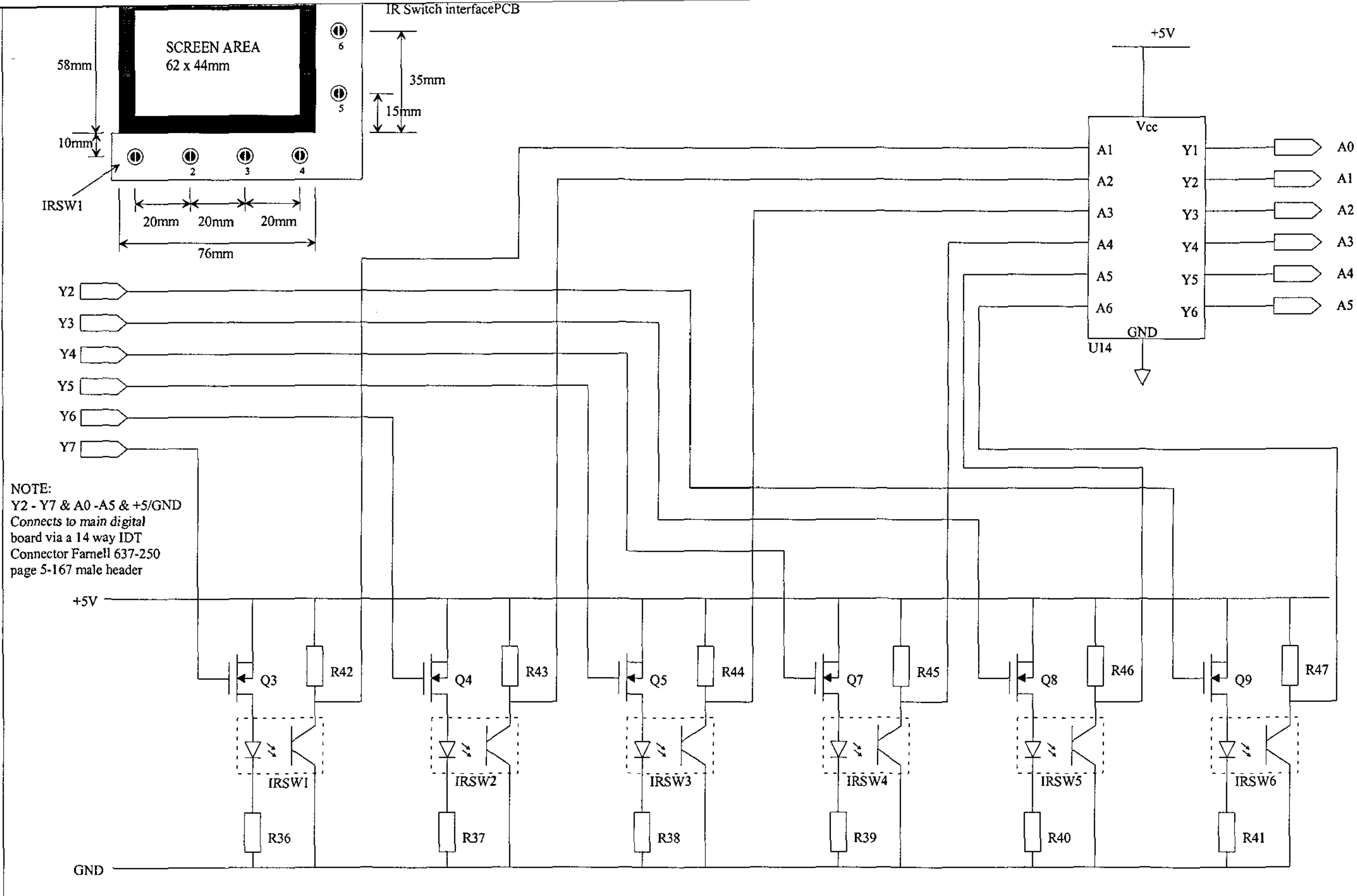
Drawn by: Darryl Newborough



Date: 06/06/00 Page 2 of 2

Input Switches

Drawn by: Darryl Newborough

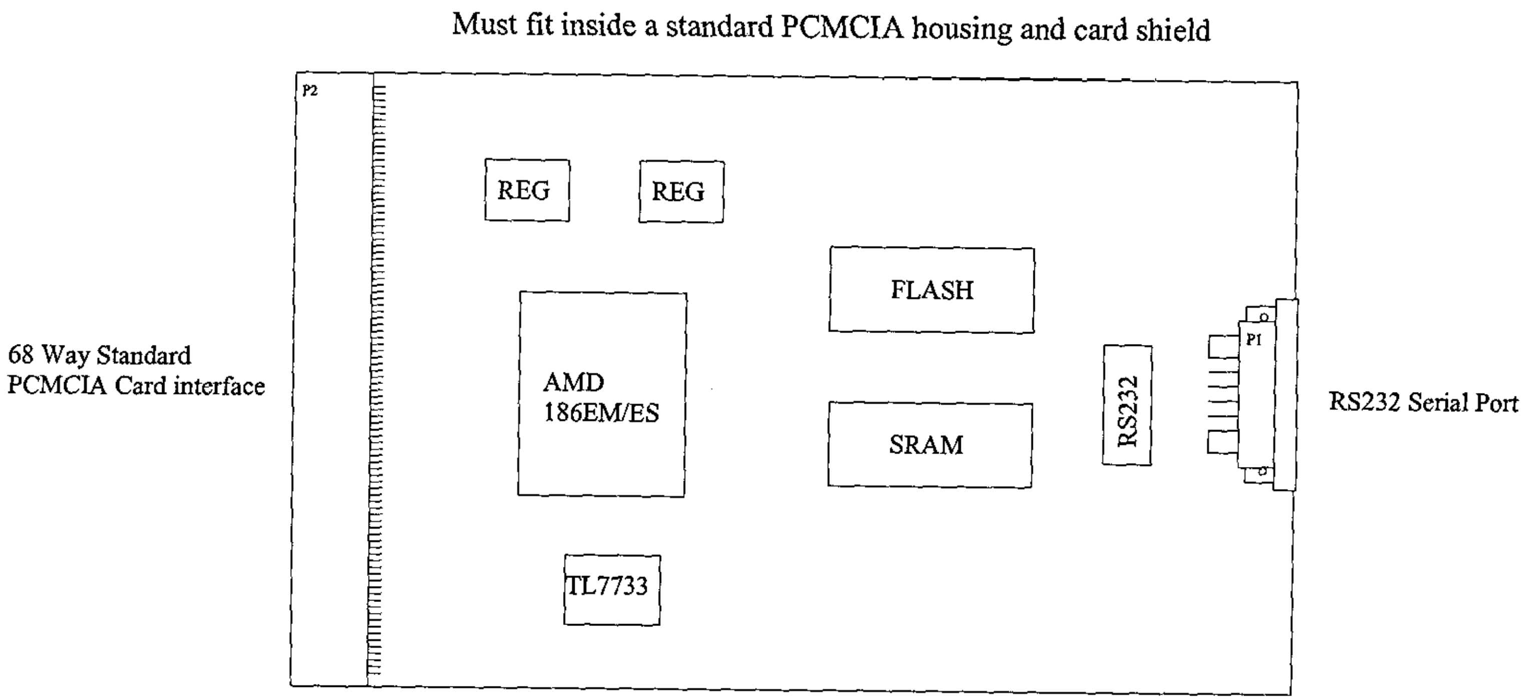


Date: 06/06/00 Page 1 of 2 IR Input Switches Drawn by: Darryl Newborough

AMD Processor board in a PCMCIA card slot package

Drawing List

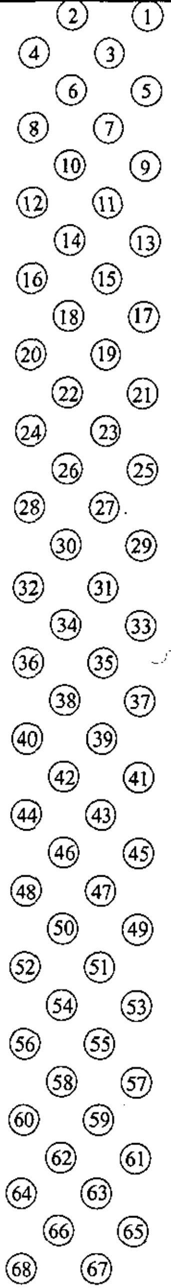
Version	Title	Date
2.0	AMD board Design	22/5/00
1.1	AMD Processor, PCMCIA Interface connections	18/04/00
2.0	AMD processor board pull-ups	22/5/00
2.0	AMD processor ES/EM Power Decoupling	22/5/00
1.0	AMD Processor board RS232 Interface	18/04/00
2.0	AMD processor board FLASH	22/5/00
2.0	AMD processor board SRAM	22/5/00
1.0	AMD processor board Regulators	18/04/00
2.0	AMD processor board Reset Module	22/05/00
	AMD demo board drawing of Microprocessor and clock	07/02/96
2.1	AMD processor board 5V version component list	18/04/00

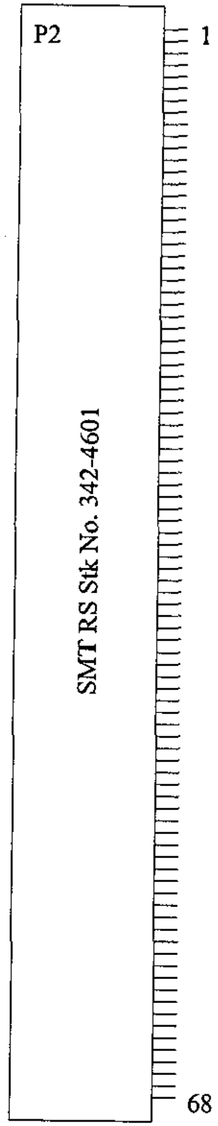


NOTE:

Layout of component for diagram purposes only

Date: 22/05/00 Version: 2.0	Drawn by: Darryl Newborough
Title: AMD board design	

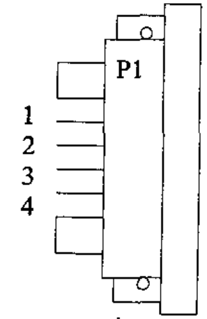




1	Vs	35	RXD1/PIO28
2	GND	36	WLB
3	D0	37	WHB
4	D1	38	/WR
5	D2	39	/RD
6	D3	40	LCS/ONCE0
7	D4	41	UCS/ONCE1
8	D5	42	INT4/PIO30
9	D6	43	INT2/PIO31
10	D7	44	/REMRST
11	D8	45	PCS5
12	D9	46	PCS6
13	D10	47	S6/LOCK/CLKD2/PIO29
14	D11	48	A1
15	D12	49	A2
16	D13	50	A3
17	D14	51	A4
18	D15	52	A5
19	INT0	53	A6
20	TMRIN1	54	A7
21	TMROUT1	55	A8
22	TMROUT0/PIO10	56	A9
23	TMR0IN0/PIO11	57	A10
24	DRQ0/INT5/PIO12	58	A11
25	DRQ1/INT6/PIO13	59	A12
26	MCS0/PIO14	60	A13
27	MCS1/PIO15	61	A14
28	PCS0/PIO16	62	A15
29	PCS1/PIO17	63	A16
30	SCLK/PIO20	64	A17/PIO7
31	SDATA/PIO21	65	A18/PIO8
32	SDEN0/PIO22	66	A19/PIO9
33	SDEN1/PIO23	67	Vs
34	TXD1/PIO27	68	GND

Card Insert Direction ←

SMT RS Stk No. 342-4825



1	GND
2	Vcc
3	TX
4	RX

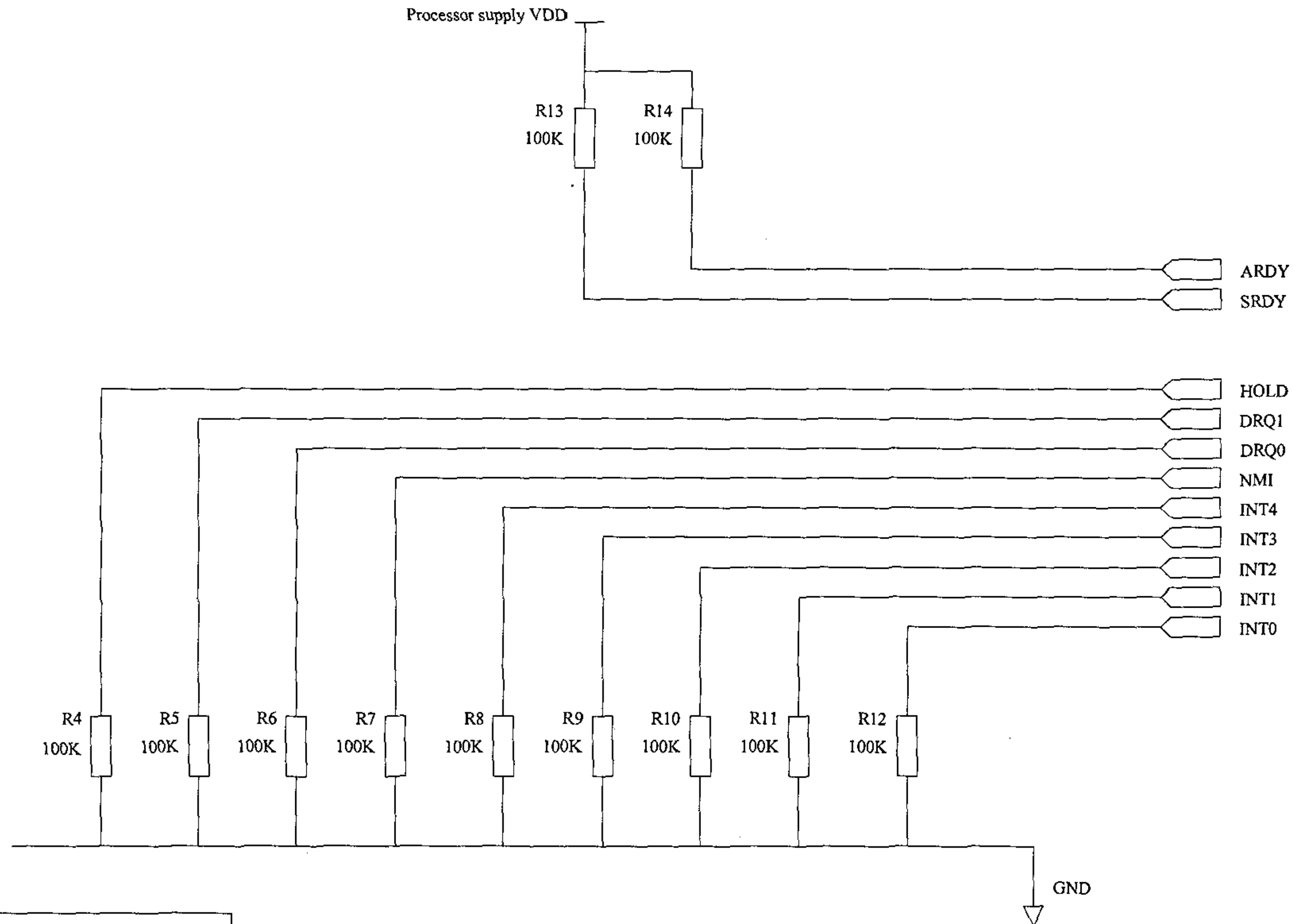
NOTES

Both connectors viewed from above.

Date: 18/04/00
Version: 1.1

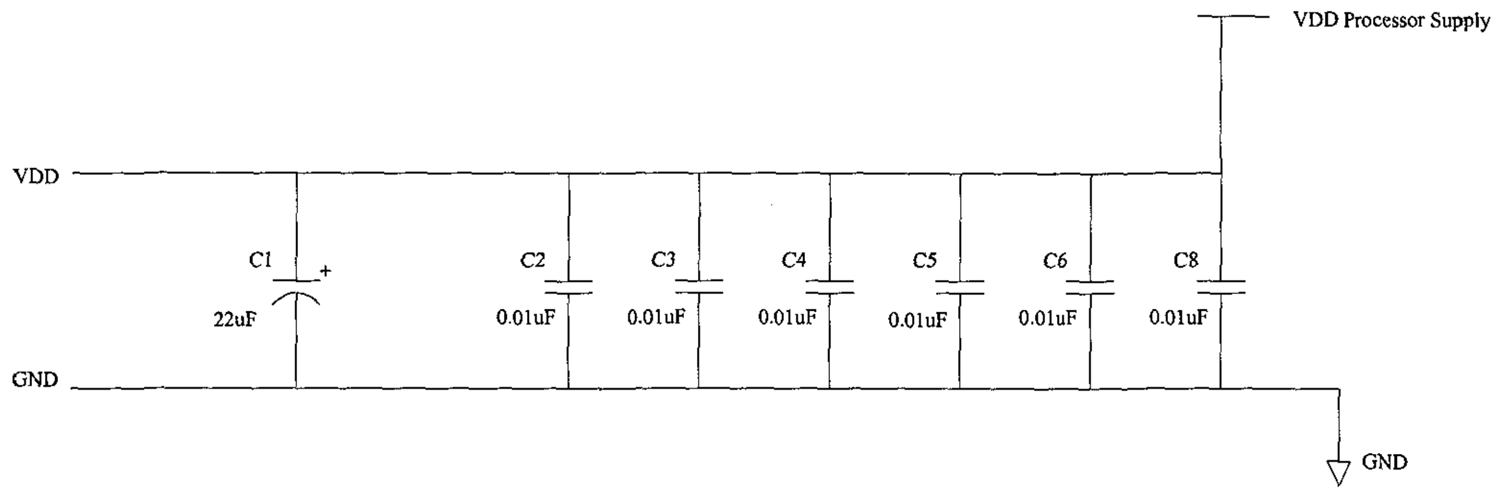
Drawn by: Darryl Newborough

Title: AMD Processor, PCMCIA interface connections



NOTE:

Date: 22/05/00 Version: 2.0	Drawn by: Darryl Newborough
Title: AMD processor board pull-ups	

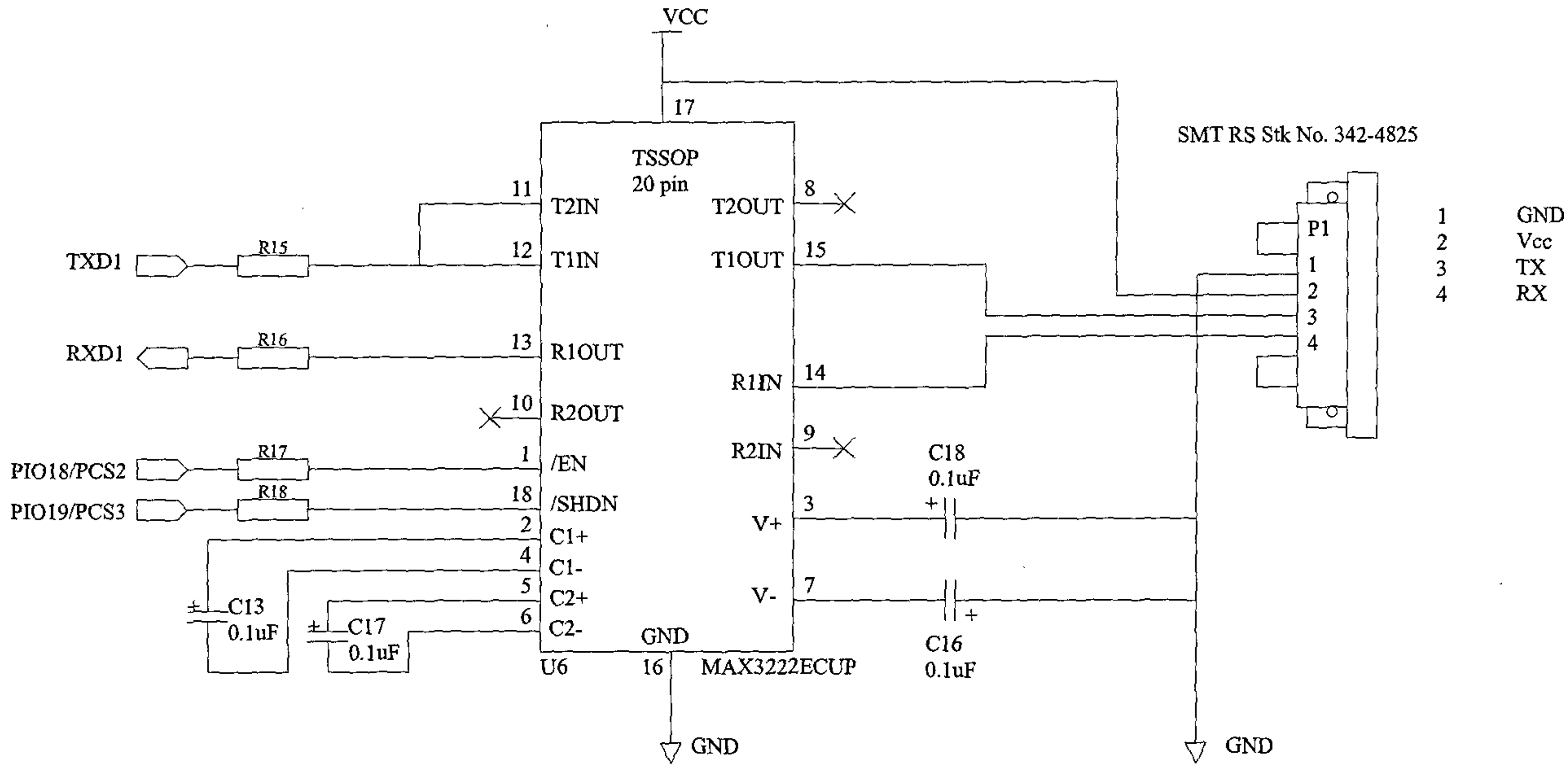


NOTE:

Date: 22/05/00
Version: 2.0

Drawn by: Darryl Newborough

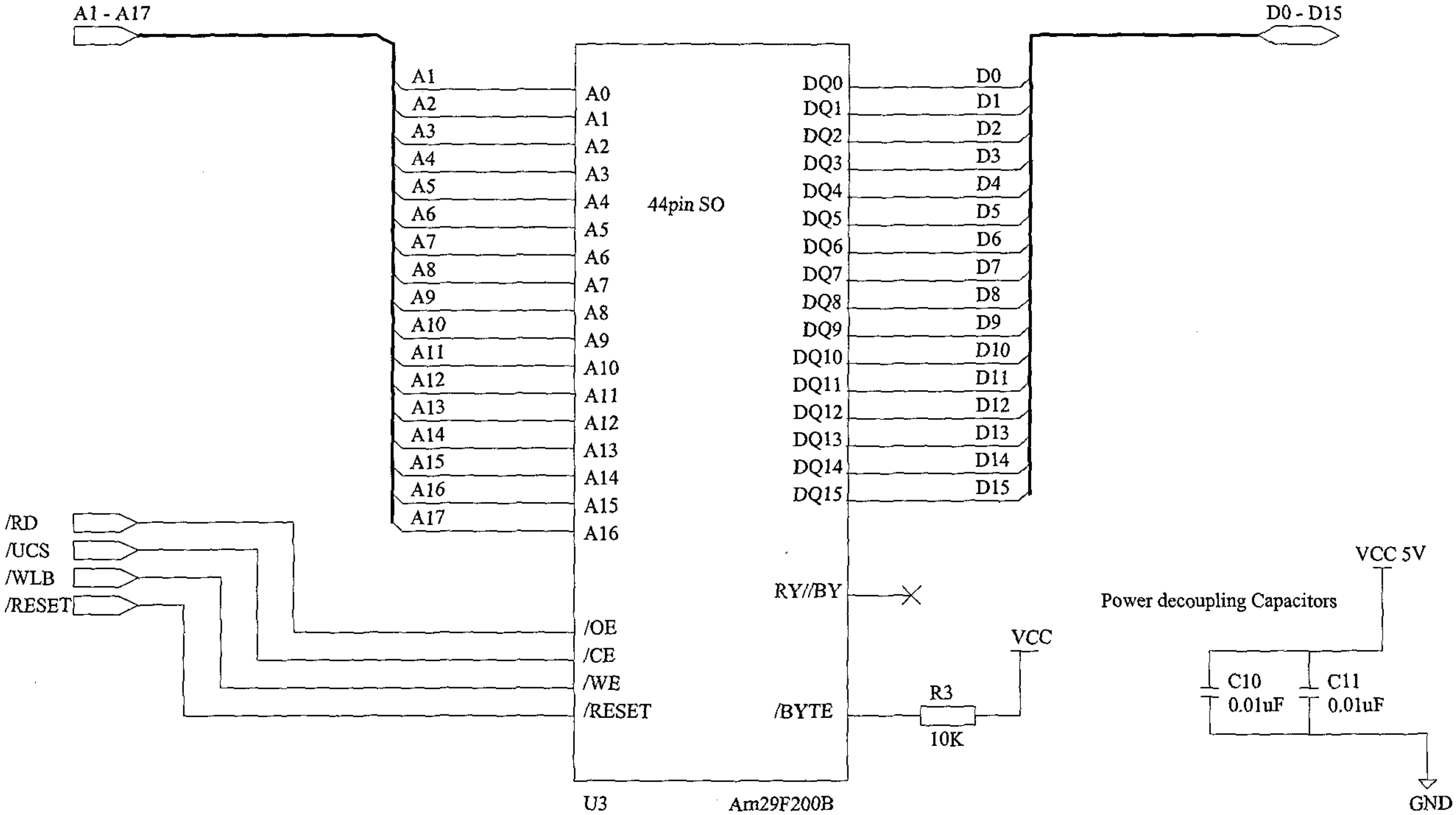
Title: AMD processor ES/EM Power Decoupling



NOTE:

MAX3222 supply range 3V - 5.5V powered via VCC

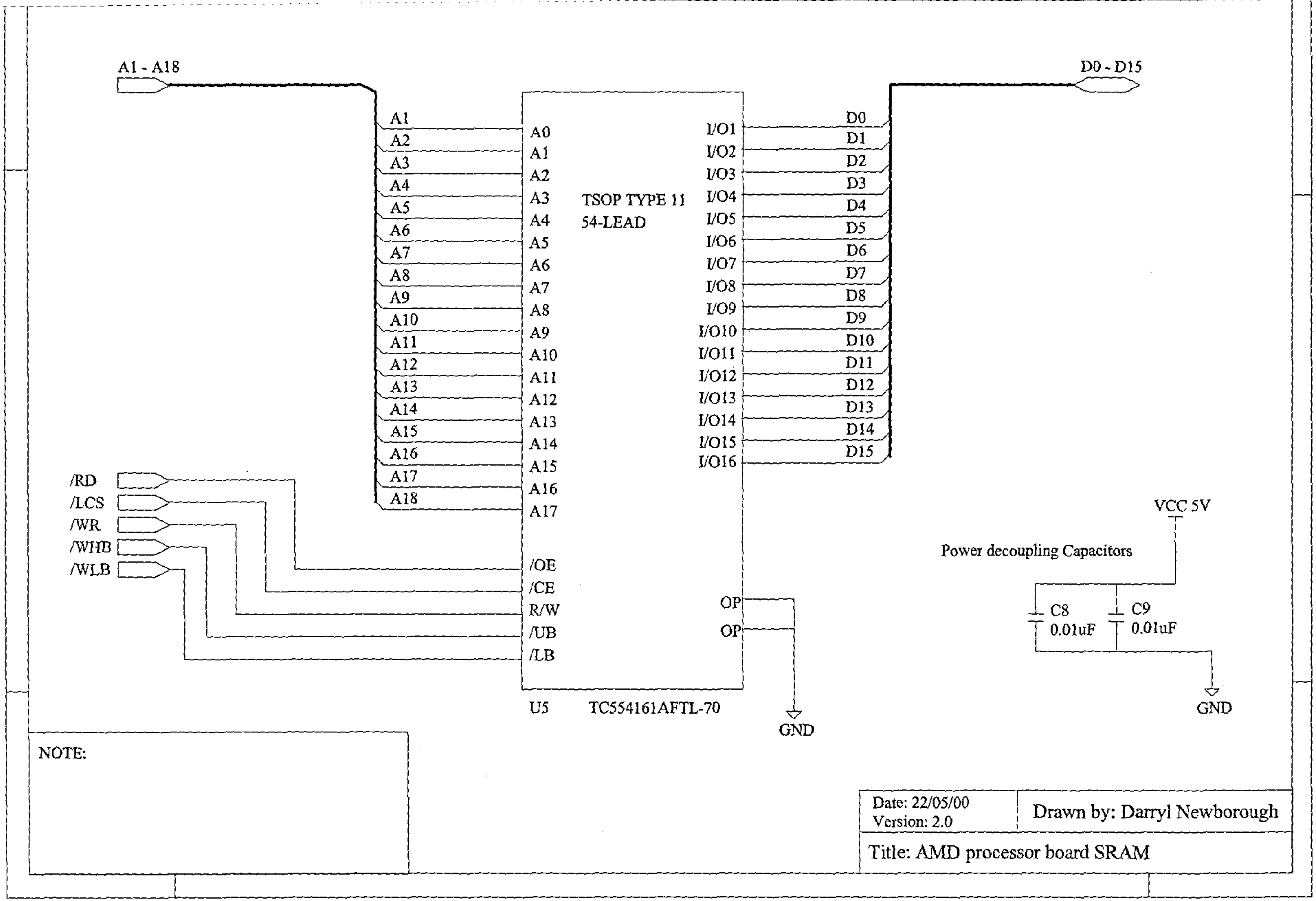
Date: 18/04/00 Version: 1.0	Drawn by: Darryl Newborough
Title: AMD processor board RS232 Interface	

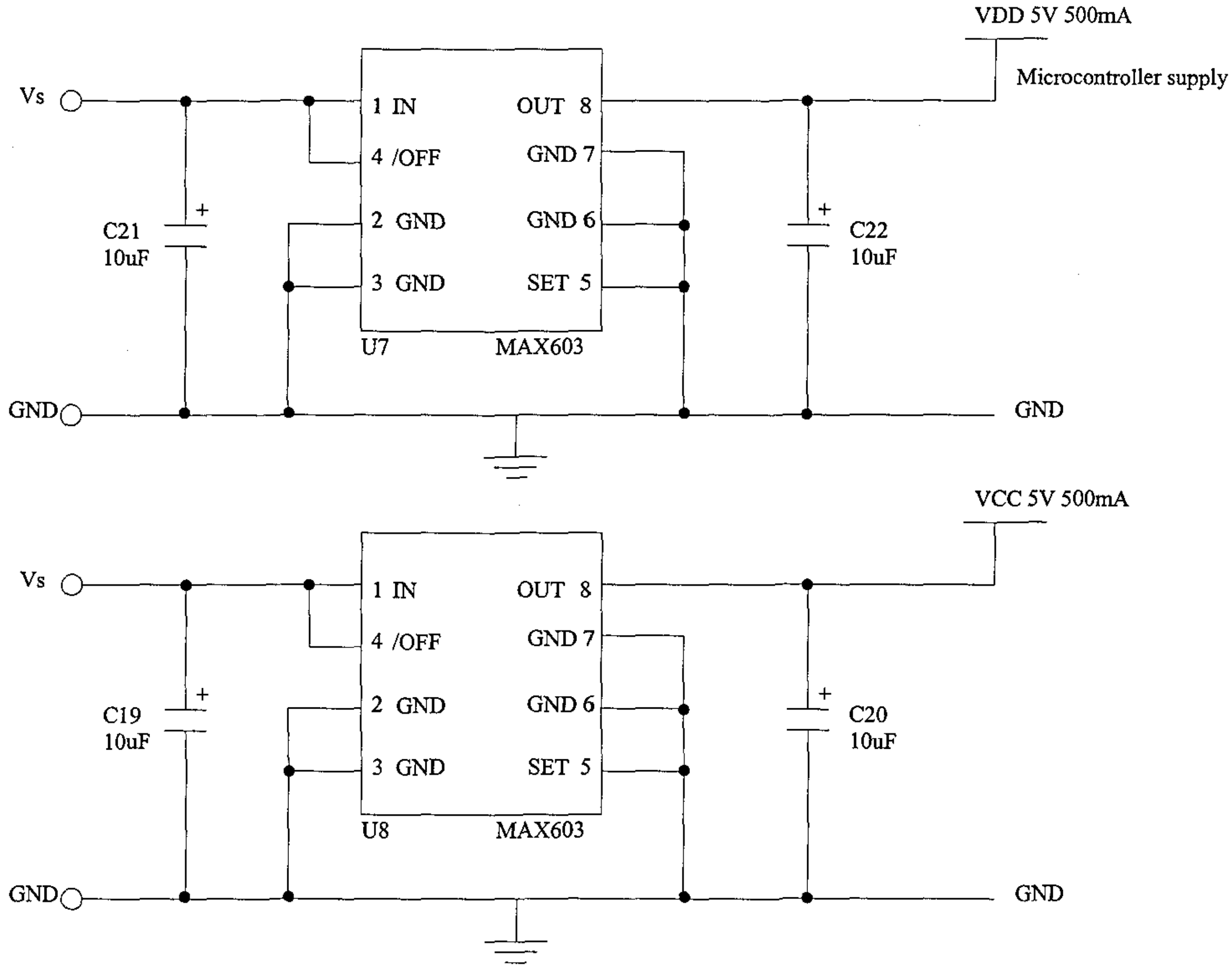


NOTE:

This part is instead of parts U3 and U4

Date: 22/05/00 Version: 2.0	Drawn by: Darryl Newborough
Title: AMD processor board FLASH	

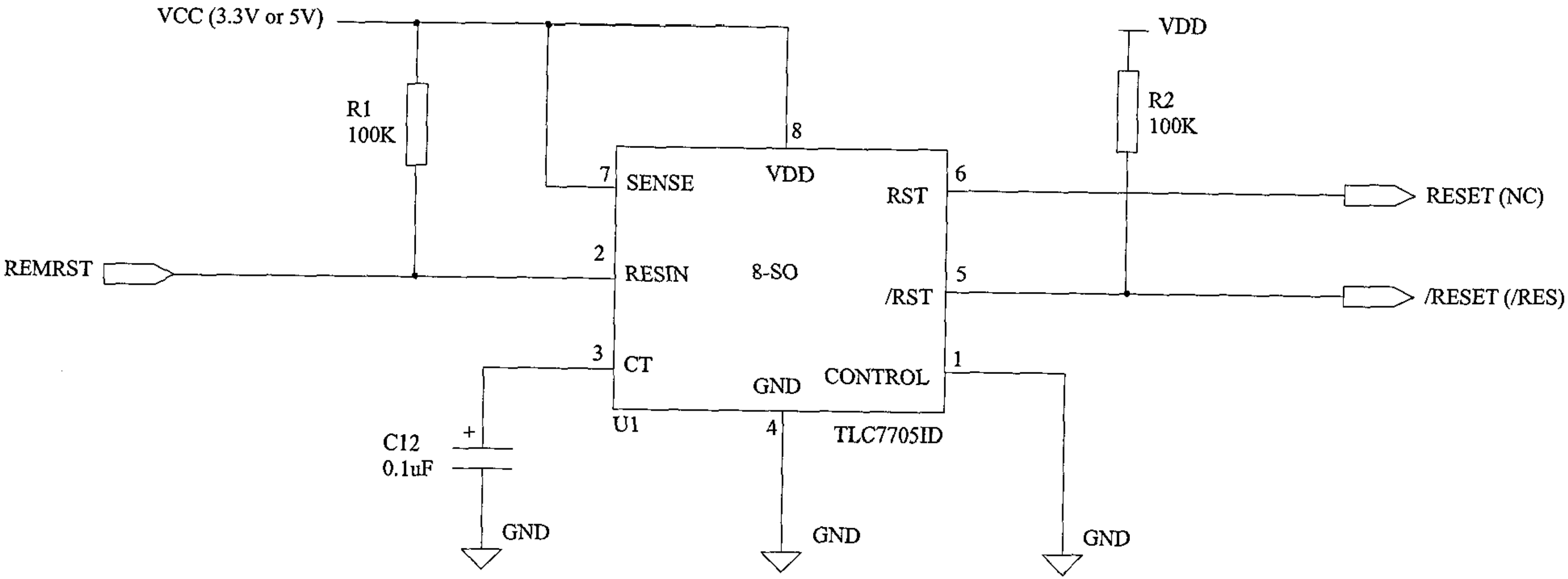

A1 - A18
D0 - D15
A1
A2
A3
A4
A5
A6
A7
A8
A9
A10
A11
A12
A13
A14
A15
A16
A17
A18
A0
A1
A2
A3
A4
A5
A6
A7
A8
A9
A10
A11
A12
A13
A14
A15
A16
A17
TSOP TYPE 11
54-LEAD
I/O1
I/O2
I/O3
I/O4
I/O5
I/O6
I/O7
I/O8
I/O9
I/O10
I/O11
I/O12
I/O13
I/O14
I/O15
I/O16
D0
D1
D2
D3
D4
D5
D6
D7
D8
D9
D10
D11
D12
D13
D14
D15
/RD
/LCS
/WR
/WHB
/WLB
/OE
/CE
R/W
/UB
/LB
OP
OP
U5
TC554161AFTL-70
GND
VCC 5V
Power decoupling Capacitors
C8
0.01uF
C9
0.01uF
GND
NOTE:
Date: 22/05/00
Version: 2.0
Drawn by: Darryl Newborough
Title: AMD processor board SRAM



NOTE

VS = Supply Voltage PCMCIA connector Pin 1, 67
GND = Ground PCMCIA conector Pin 2, 68
VDD = 5V supply for the Processor
VCC = 5V supply for the FLASH and SRAM etc.

Date: 22/05/00 Version: 2.0	Drawn by: Darryl Newborough
Title: AMD processor board Regulators	



NOTE:

Date: 22/05//00
Version: 2.0

Drawn by: Darryl Newborough

Title: AMD processor board Reset Module

AMD Processor board in a PCMCIA card slot package **5V Version 2.1**

Qty	Ref	Description	MFG1	Part No.	Supplier	Price	Price +25
1	U1	Reset Controller, SMT	TI	TL7705ACD	In Stock		
1	U2	AMD Microprocessor 5V	AMD	AMD186EM/ES-40KC/W	In Stock		
1	U3	128K x 16 Flash EPROM (SMT)	AMD	Am29F200AT-70SC	In Stock		
1	U5	256K x 16 Toshiba SRAM	Toshiba	TC554161AFT-70L	In Stock		
1	U6	RS232 Driver, 20 TSSOP(SMT)	Maxim	MAX3222ECUP			
1	Y1	Crystal, 40MHz High Freq. 18pF					
2	U7-U8	Linear Reg. Low-drop out 500mA 5V	Maxim	MAX603CSA			
1	P1	4 way I/O shielded		342-4825	RS		
1	P2	68 way std, SMT (option Std or Cardbus)		342-4601	RS		
1	C1	22uF, SMT, 16V					
10	C2-C11	0.01uF, SMT 0603					
5	C12, C13, C16-C18	0.1uF					
2	C14, C15	15pF, SMT, 0603					
4	C19-C22	10uF, SMT, Tant, 16V					
13	R1, R2, R4 - R14	100K 0603 Tolerance 1%					
1	R3	10K 0603 Tolerance 1%					
4	R15 -R18	0R Jumpers					

APPENDIX B

EXTENDED COMMANDS

1. APPENDIX B: EXTENDED COMMANDS

1.1 Commands

1. Reset Command (Global) (0xFF)

   The global reset command causes all unit that receive the command to enter NDM and as if initial powered up

2. Release Command (Cannot be used globally) (0xAA) x 3

   Operates the release mechanism that releases the positively buoyant transponder and float to the surface. This command must be received three times with five seconds of the first command for it to activate the release mechanism.

3. Co-ordinate Transfer command (0x20)

   This command is transmitted a short period before a *data3_int* is transmitted by the master globally conveying positional information of the seabed transponders. Receiving the extended command readies the slave units for a data packet that will contain the co-ordinates of the *from_id* transponder. A slave unit must receive the data packet within a predetermined time period or it will respond to the co-ordinate transfer command to indicate that it did not receive the co-ordinate data, co-ordinate data error response.

4. Change response time (0x30)

   This instructs a certain unit (identified) or all units to change the response dead time; the 8-bit data allows a response time between 50ms to 1sec to be set.

5. Change Data Rate (0x40)

   Used to change the default modulation scheme, to enable higher data rates to be achieved.

6. Emergency (0x50)

   When the emergency command is received general position fixing is reduce and the emergency transponder is given higher priority.

7. Fast position fixing 1 (0x60)

   This enable a single transponder to transmit a number of ranging commands, which enable the transponder to have a fast position update rate.

8. Fast position fixing 2 (0x70)

   Fast positioning fixing 2 enables all the mobile units to position themselves using a replay ranging technique.

9. Passive Position (0x80)

   Enables and disables passive positioning of acoustic maker beacons

10. Relinquish Master Role (0x90)

    Issued by a user, i.e. the surface monitoring unit or a diver unit, which allow a master transponder to enter the NDM and waits to be discovered by the new master.

11. Assign Master Role (0xA0)

12. Change Output Power (0xB0)

    The following byte is the output digitally controlled potentiometer value.

13. Test Mode (0xC0)

    Bit Error Rate test, position accuracy, cable calibration, acoustic calibration. The acoustic calibration can be used to calculated the speed of sound if an accurate hydrophone seperation if known.

14. Status Command (0xD0)

    Request the a status response from the addressed transponder. This command cannot be used globally.

15. Get Maximum Data Packet Length (0xE0)

    The Master transponder resquest the maximum packet length that can be used to communicate with a particular transponder.

16. Maximum Data Packet Length Response (0xF0)

    Slave response to command 0xE0, one data byte returned, 6 bit indicating the maximum length, from 8 – 72-bits and one bit indicate amplitude of reverberation following the direct signal and one bit to indicate if an exact arrival time measurement was possible.

17. Coordinated data response (0x02)

Following this command are the x, y, z co-ordinates of the responding transponder.

18. Status Response (0x03)

The current status of the transponder is transmitted, this includes:mode, output power, noise level, inter-packet time, deadtime, memory status, number of connected transponders, battery status etc.

APPENDIX C

POSITIONING

1. APPENDIX C: POSITIONING

```
/*****************************************************************
* MatrixInversion: take two 3x3, 2D double arrays and calculates    *
* the inverse of m[3][3] and places the result in im[3][3]          *
* fail is matrix is singular                                         *
*******************************************************************/

#include <math.h>
#include <stdio.h>
#include <kernel.h>

void MatrixInversion(double m[3][3], double im[3][3])
{
	double det;
	double val;
	int flag = 0;

	// x y z
	// p q r
	// a b c

 /*calculate determinant*/
	det=m[0][0]*((m[1][1]*m[2][2])-(m[2][1]*m[1][2]))-m[0][1]*((m[1][0]*m[2][2])
	-(m[2][0]*m[1][2]))+m[0][2]*((m[1][0]*m[2][1])-(m[1][1]*m[2][0]));

	if(det < 0.00001 && det > -0.00001)	/*if matrix is singular program terminates*/
	{
		printk(" Error");
	}

	val=1/det;

	im[0][0]=((m[1][1]*m[2][2])-(m[2][1]*m[1][2]));	/*calculate cofactors*/
	im[0][1]=((-1)*((m[1][0]*m[2][2])-(m[2][0]*m[1][2])));
	im[0][2]=((m[1][0]*m[2][1])-(m[1][1]*m[2][0]));
	im[1][0]=((-1)*((m[0][1]*m[2][2])-(m[2][1]*m[0][2])));
	im[1][1]=((m[0][0]*m[2][2])-(m[2][0]*m[0][2]));
	im[1][2]=((-1)*((m[0][0]*m[2][1])-(m[2][0]*m[0][1])));
	im[2][0]=((m[0][1]*m[1][2])-(m[1][1]*m[0][2]));
	im[2][1]=((-1)*((m[0][0]*m[1][2])-(m[1][0]*m[0][2])));
	im[2][2]=((m[0][0]*m[1][1])-(m[1][0]*m[0][1]));


		#ifdef __PRINT_STATS
	printk("\nThe inverse of the given matrix is: ");
	printk("%2f\n\r", det);		/*absolute value of determinant*/

	printk("%2f, %2f, %2f\n\r", im[0][0], im[1][0], im[2][0]);
	printk("%2f, %2f, %2f\n\r", im[0][1], im[1][1], im[2][1]);
	printk("%2f, %2f, %2f\n\r", im[0][2], im[1][2], im[2][2]);

	printk("\nor\n");
	im[0][0]*=val;im[0][1]*=val;im[0][2]*=val;	/*multiply each value by determinant*/
	im[1][0]*=val;im[1][1]*=val;im[1][2]*=val;
	im[2][0]*=val;im[2][1]*=val;im[2][2]*=val;

	printk("%2f, %2f, %2f\n\r", im[0][0], im[1][0], im[2][0]);
	printk("%2f, %2f, %2f\n\r", im[0][1], im[1][1], im[2][1]);
	printk("%2f, %2f, %2f\n\r", im[0][2], im[1][2], im[2][2]);
	#endif
}
```

1.1 Performance Profiling

Figure C.1 shows the average time in milliseconds to perform various mathematical functions include a 3x3 matrix inverse shown in the above C code. All functions shown are operating at double precision, hence 64-bit.

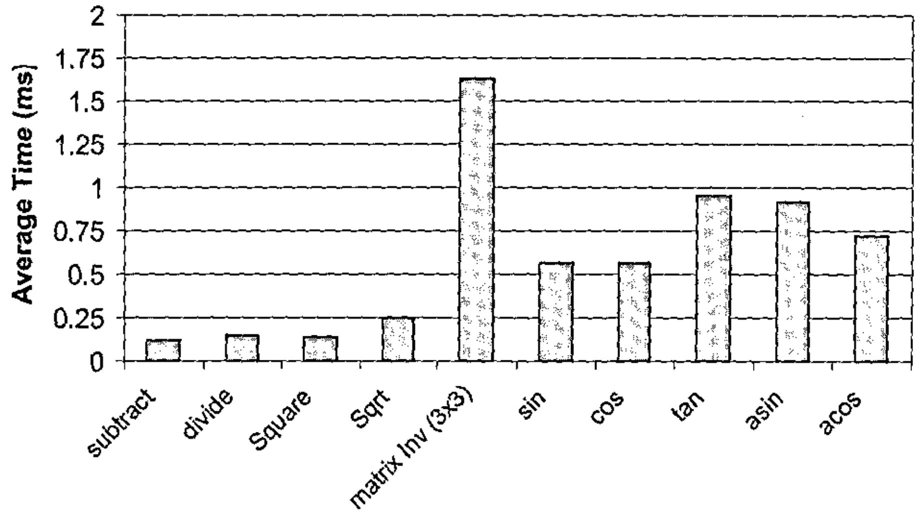


Figure C.1 Processor Profiling